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Soli et al.

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[54] **AUDITORY PROSTHESIS, NOISE SUPPRESSION APPARATUS AND FEEDBACK SUPPRESSION APPARATUS HAVING FOCUSED ADAPTIVE FILTERING**

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[51] **Int. Cl.**[6] **H04B 15/00**

[52] **U.S. Cl.** **381/94**; 381/71

[58] **Field of Search** 381/94, 71, 83, 93

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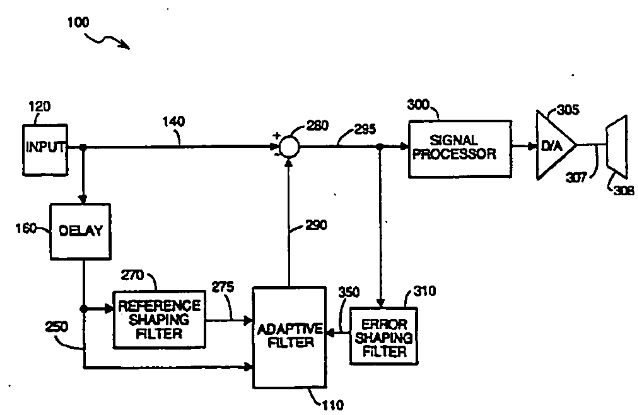
Attorney, Agent, or Firm—Gary L. Griswold; Walter N. Kirn; William D. Bauer

[57] **ABSTRACT**

A noise and feedback suppression apparatus processes an audio input signal having both a desired component and an undesired component. When implemented so as to effect noise cancellation, the apparatus includes a first filter operatively coupled to the input signal. The first filter generates a focused reference signal by selectively passing an audio spectrum of the input signal which primarily contains the undesired component. The reference signal is supplied to an adaptive filter disposed to filter the input signal so as to provide an adaptive filter output signal. A combining network subtracts the adaptive filter output signal from the input signal to create an error signal. The noise suppression apparatus further includes a second filter for selectively passing to the adaptive filter an audio spectrum of the error signal substantially encompassing the spectrum of the undesired component of the input signal. This cancellation effectively removes the undesired component from the input signal without substantially affecting the desired component of the input signal. When the present apparatus is implemented so as to suppress feedback the adaptive filter output signal is employed to cancel a feedback component from the input signal.

34 Claims, 11 Drawing Sheets



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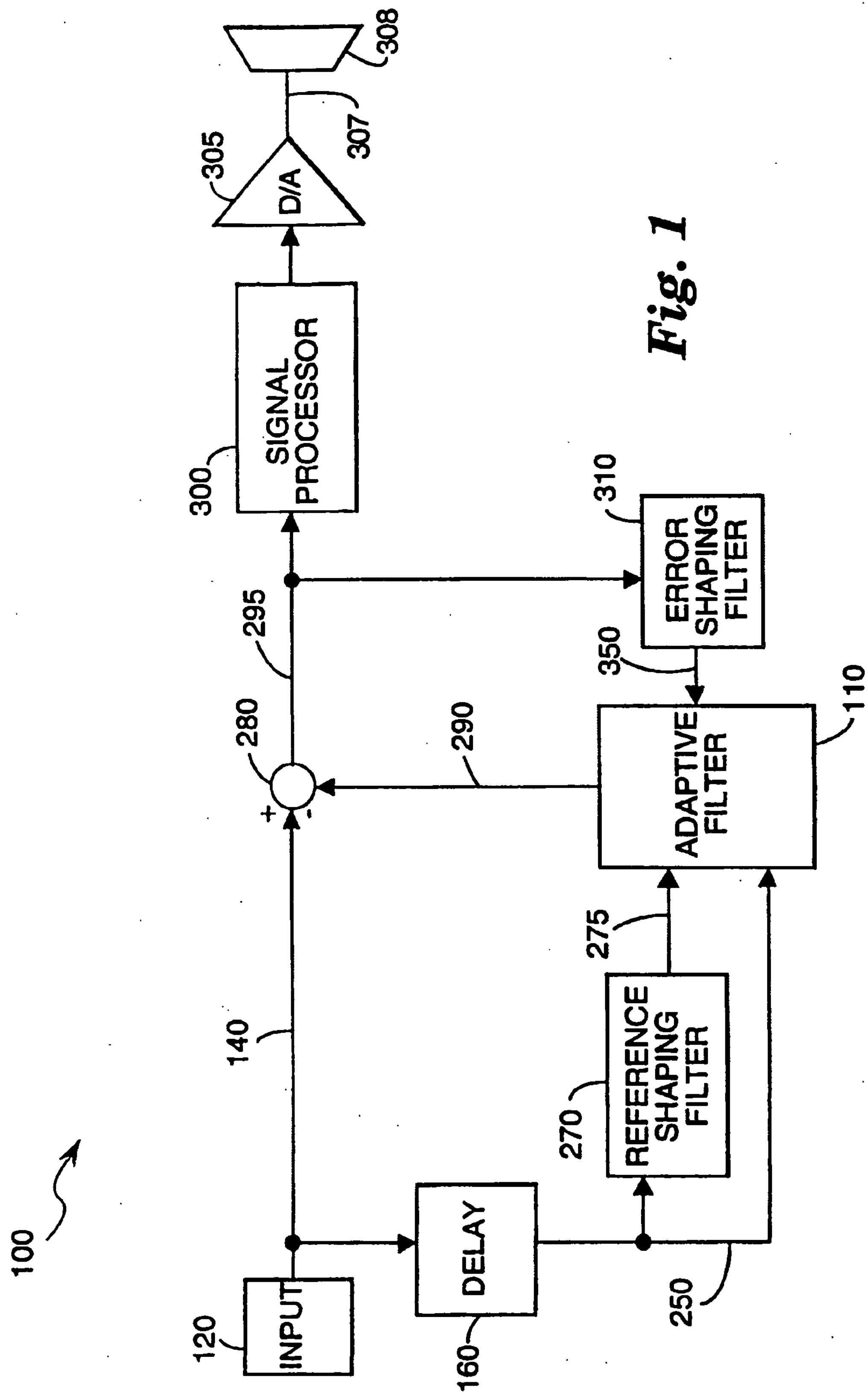

308
307
305
D/A
SIGNAL PROCESSOR
300
310
ERROR SHAPING FILTER
295
350
110
ADAPTIVE FILTER
280
290
+
-
275
REFERENCE SHAPING FILTER
270
140
100
INPUT
120
DELAY
160
250

Fig. 1

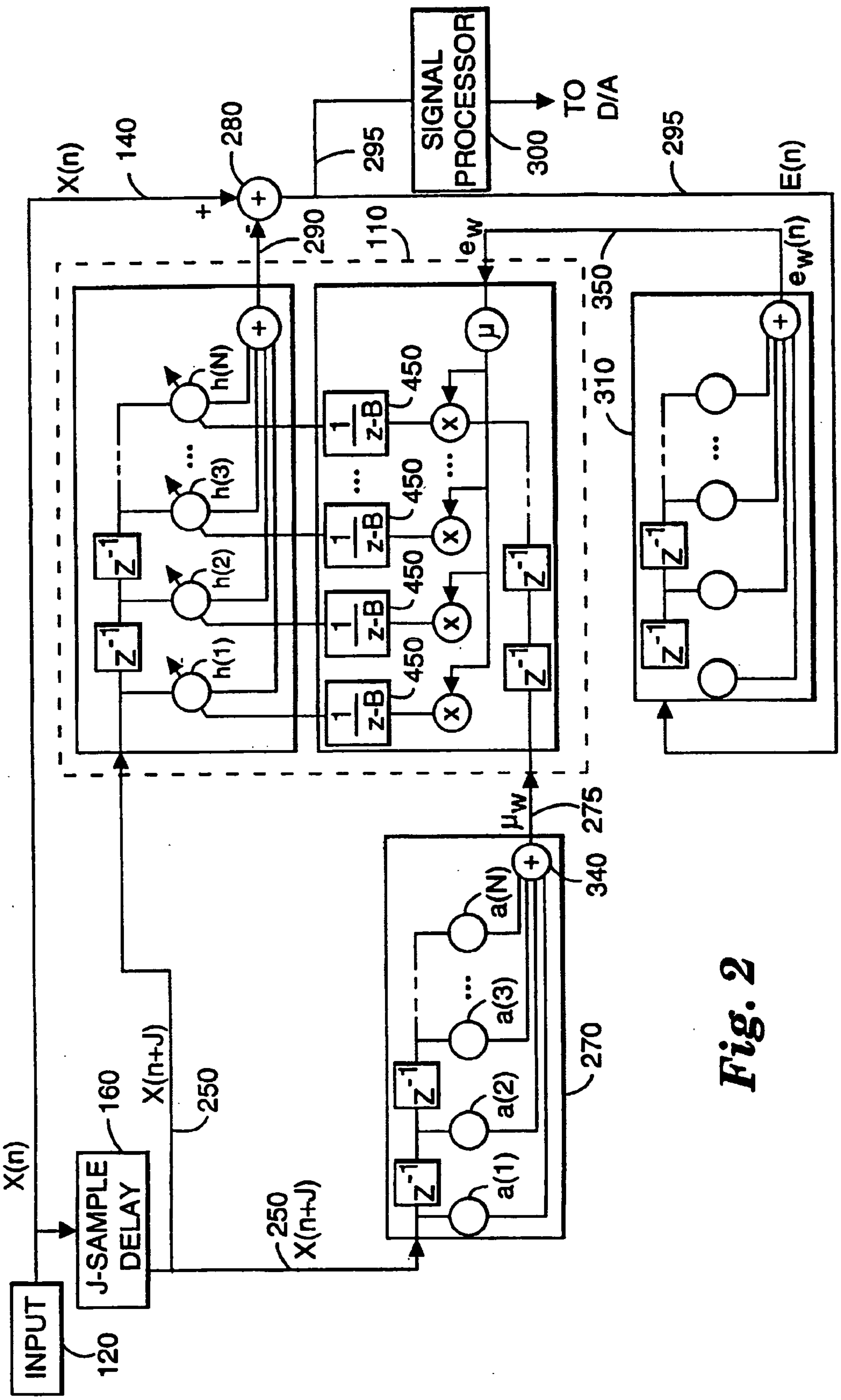

INPUT
120
J-SAMPLE DELAY
160
X(n)
140
X(n+J)
250
280
290
110
295
SIGNAL PROCESSOR
300
TO D/A
E(n)
e_W
$e_W(n)$
350
310
450
$\frac{1}{z-B}$
z^{-1}
h(1)
h(2)
h(3)
h(N)
a(1)
a(2)
a(3)
a(N)
μ
μ_W
275
340
270

Fig. 2

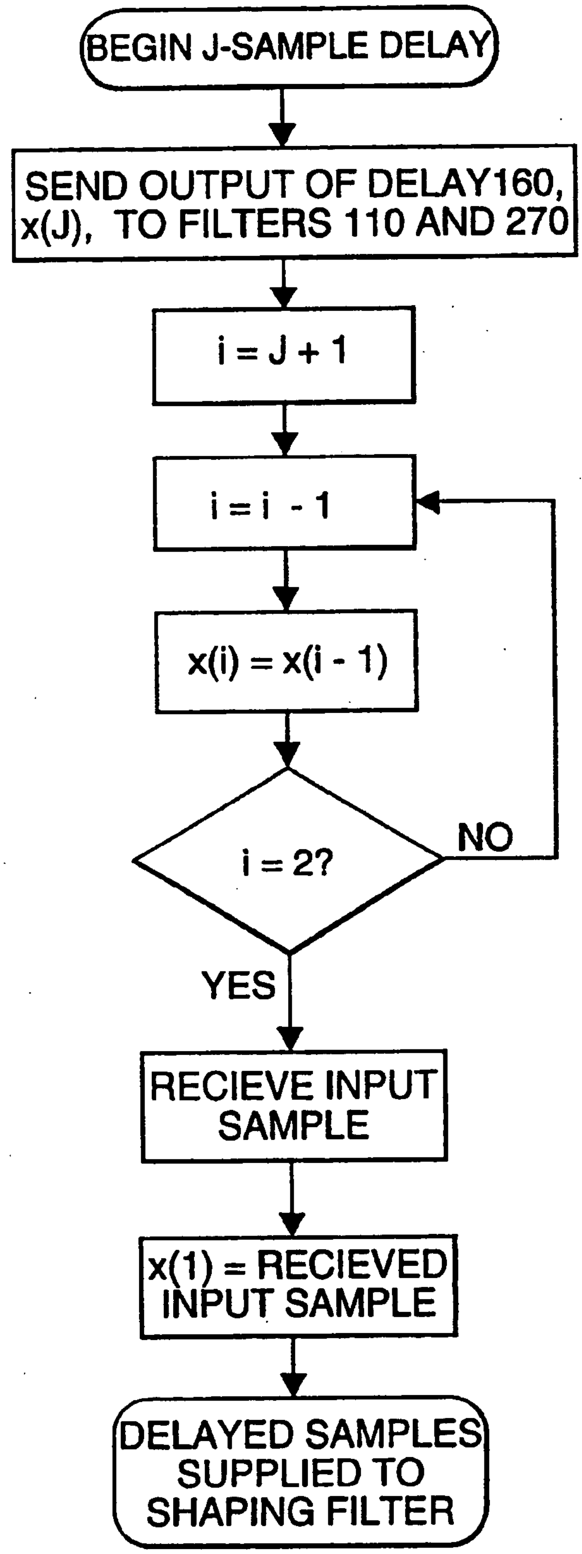

BEGIN J-SAMPLE DELAY
SEND OUTPUT OF DELAY160, x(J), TO FILTERS 110 AND 270
i = J + 1
i = i - 1
x(i) = x(i - 1)
i = 2?
NO
YES
RECIEVE INPUT SAMPLE
x(1) = RECIEVED INPUT SAMPLE
DELAYED SAMPLES SUPPLIED TO SHAPING FILTER

Fig. 3

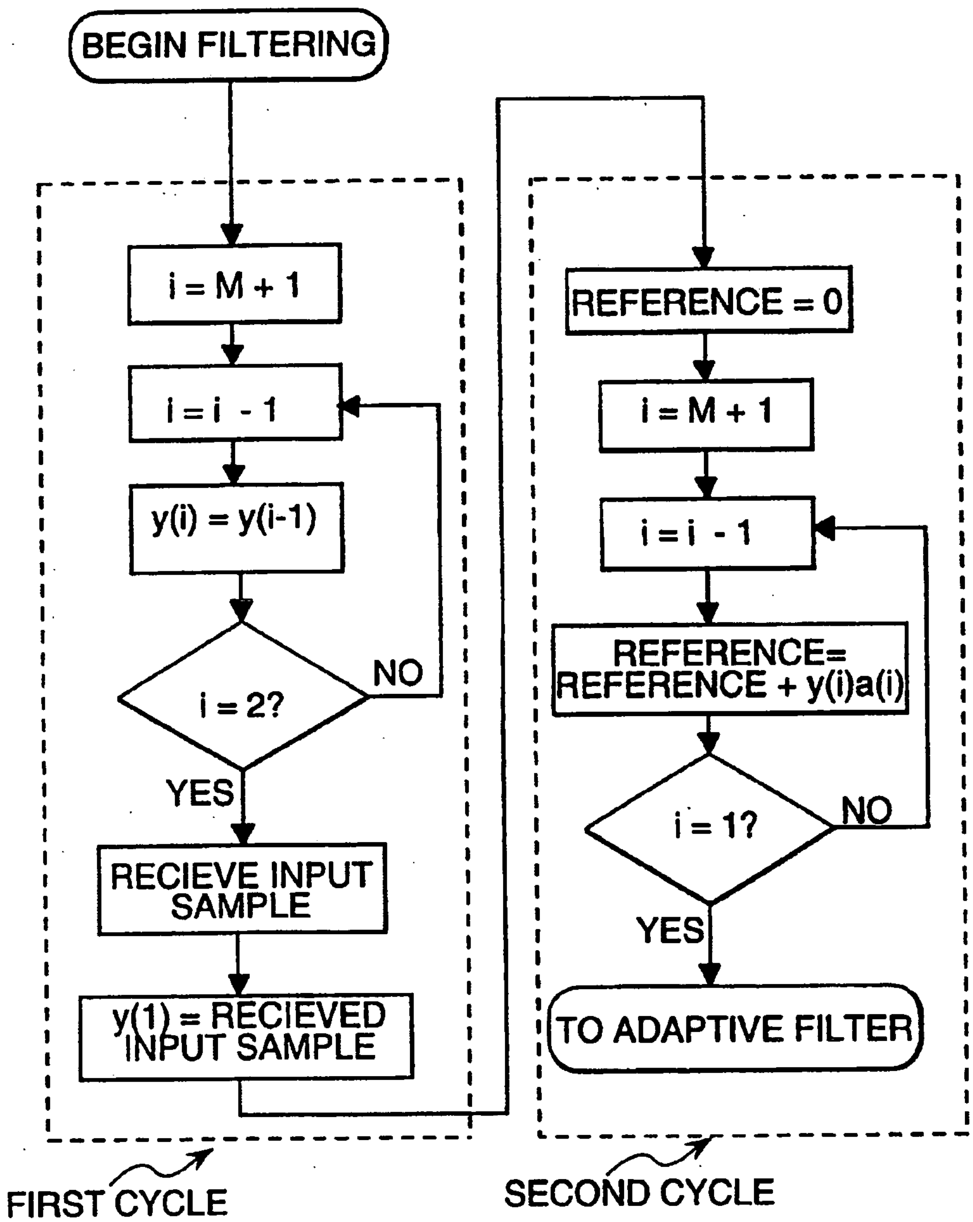

BEGIN FILTERING
i = M + 1
i = i - 1
y(i) = y(i-1)
i = 2?
NO
YES
RECIEVE INPUT SAMPLE
y(1) = RECIEVED INPUT SAMPLE
FIRST CYCLE
REFERENCE = 0
i = M + 1
i = i - 1
REFERENCE= REFERENCE + y(i)a(i)
i = 1?
NO
YES
TO ADAPTIVE FILTER
SECOND CYCLE

Fig. 4

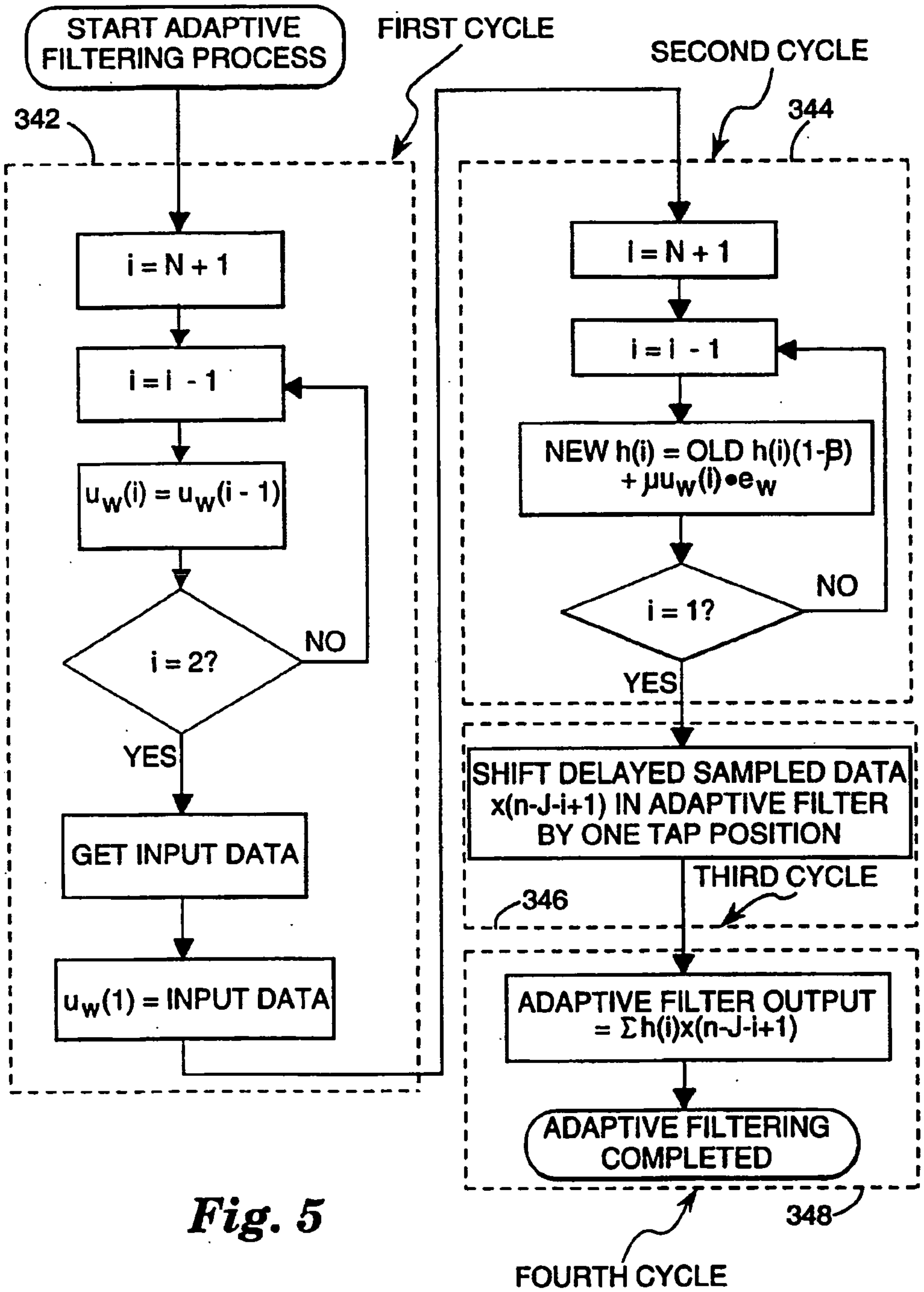

START ADAPTIVE FILTERING PROCESS
FIRST CYCLE
342
i = N + 1
i = i - 1
$u_W(i) = u_W(i - 1)$
i = 2?
NO
YES
GET INPUT DATA
$u_W(1)$ = INPUT DATA
SECOND CYCLE
344
i = N + 1
i = i - 1
NEW h(i) = OLD h(i)(1-β) + $\mu u_W(i) \bullet e_W$
i = 1?
NO
YES
SHIFT DELAYED SAMPLED DATA x(n-J-i+1) IN ADAPTIVE FILTER BY ONE TAP POSITION
THIRD CYCLE
346
ADAPTIVE FILTER OUTPUT = Σh(i)x(n-J-i+1)
ADAPTIVE FILTERING COMPLETED
348
FOURTH CYCLE

Fig. 5

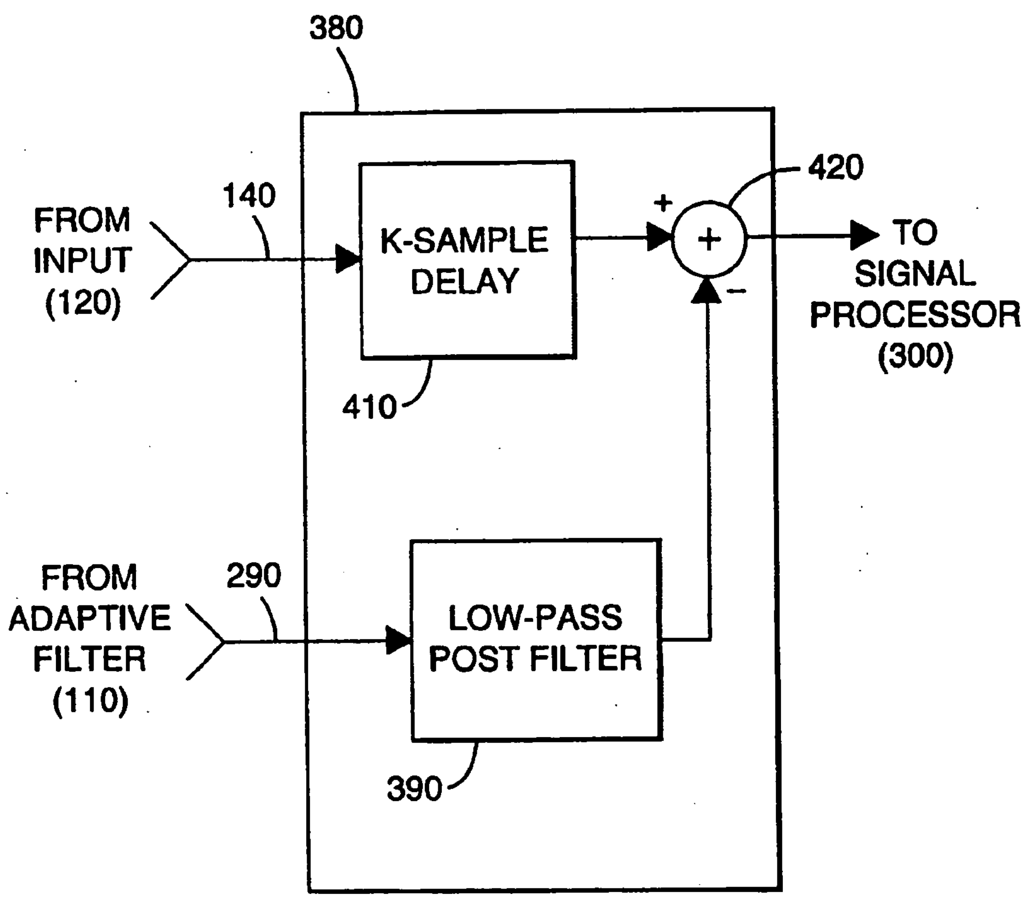

380
FROM INPUT (120)
140
K-SAMPLE DELAY
410
+
420
+
−
TO SIGNAL PROCESSOR (300)
FROM ADAPTIVE FILTER (110)
290
LOW-PASS POST FILTER
390

Fig. 6

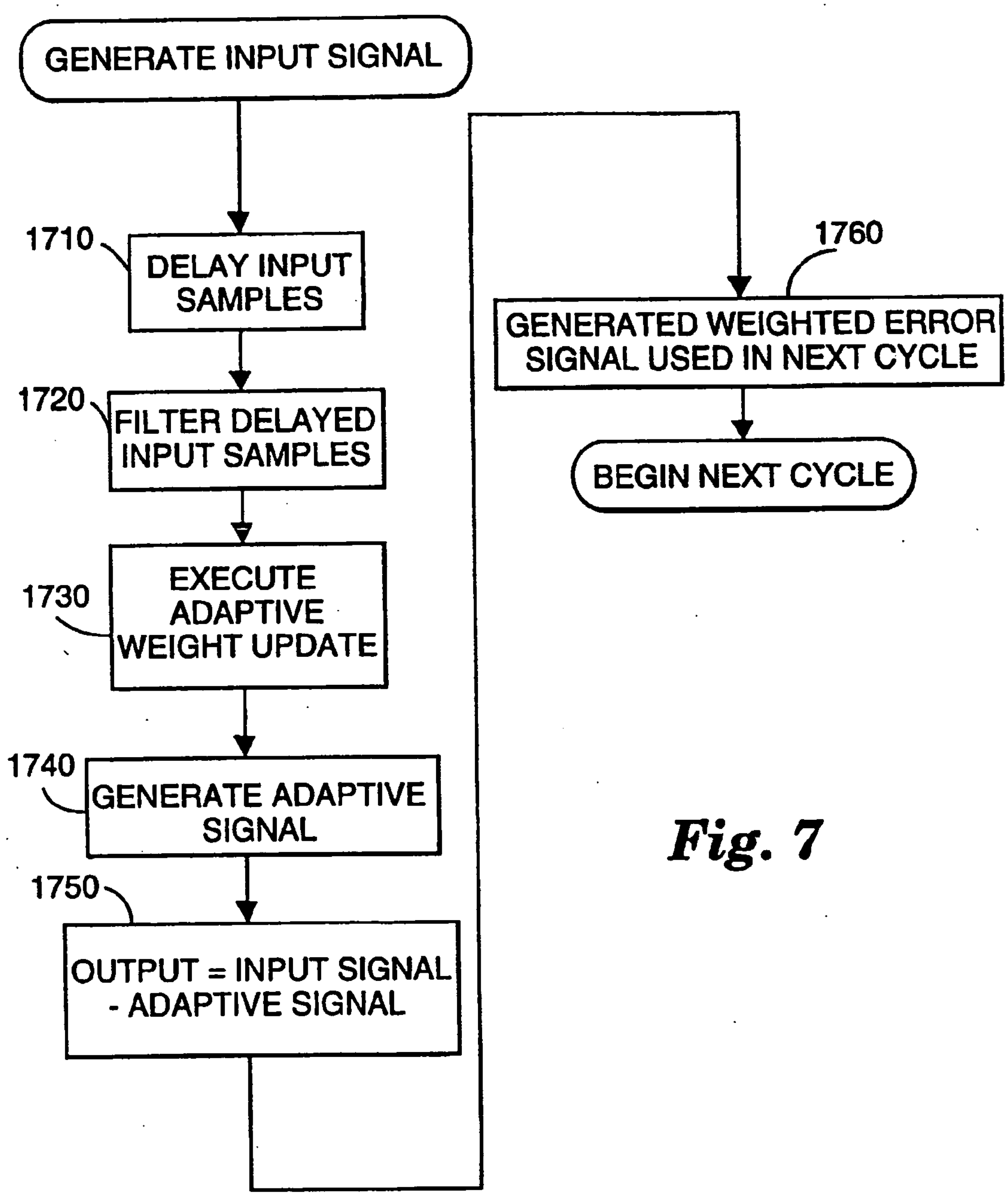
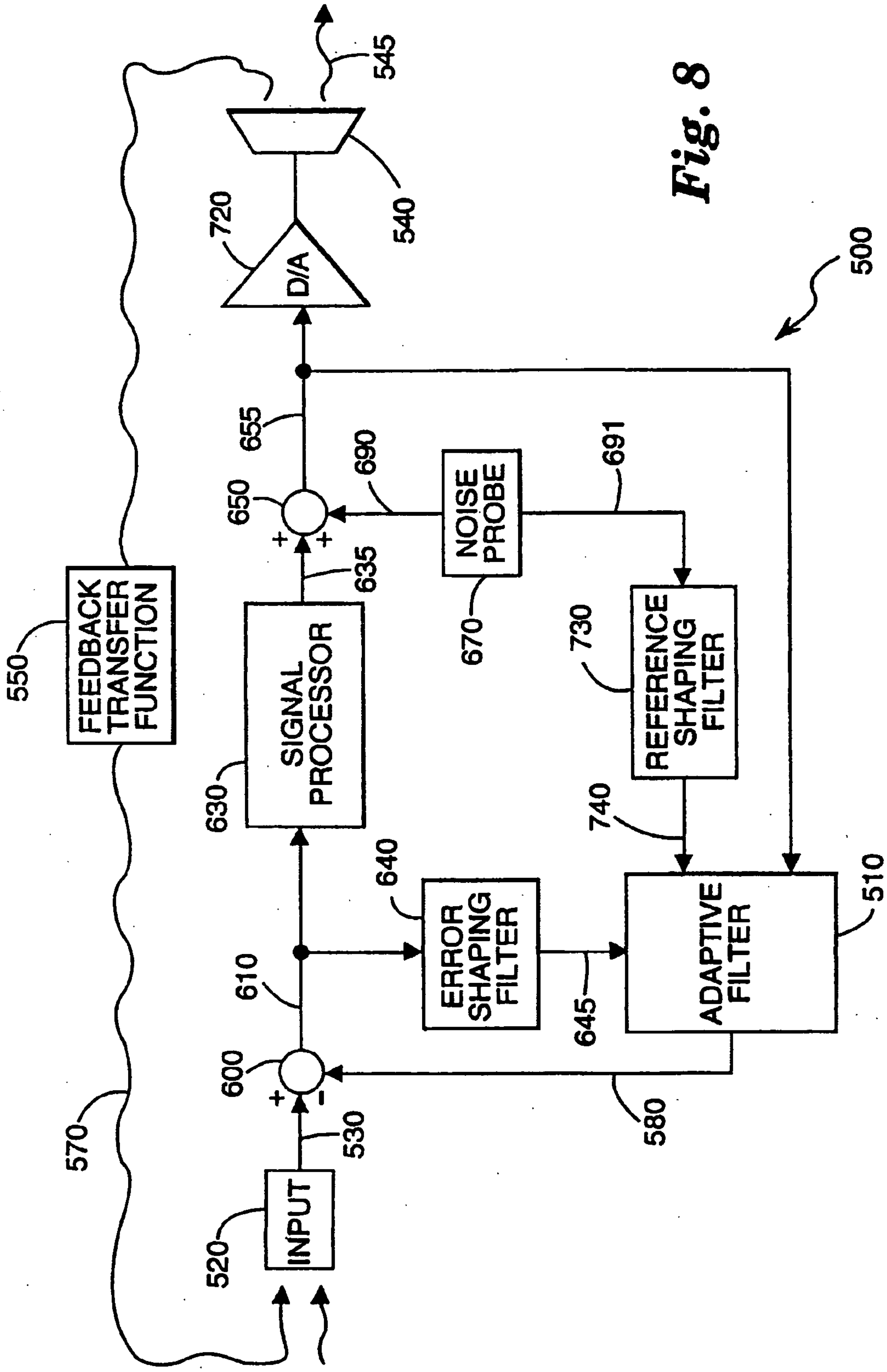
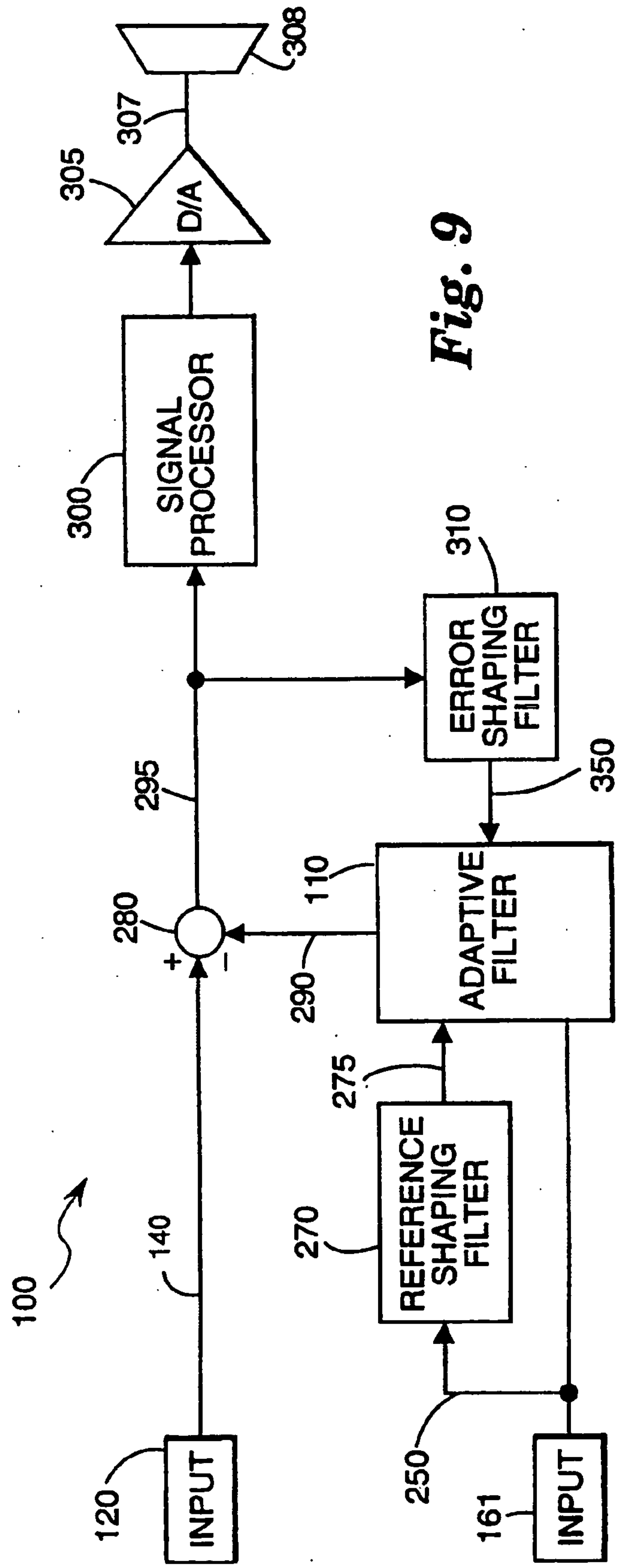

GENERATE INPUT SIGNAL
1710
DELAY INPUT SAMPLES
1720
FILTER DELAYED INPUT SAMPLES
1730
EXECUTE ADAPTIVE WEIGHT UPDATE
1740
GENERATE ADAPTIVE SIGNAL
1750
OUTPUT = INPUT SIGNAL - ADAPTIVE SIGNAL
1760
GENERATED WEIGHTED ERROR SIGNAL USED IN NEXT CYCLE
BEGIN NEXT CYCLE

Fig. 7



545
720
D/A
540
500
655
650
690
NOISE PROBE
691
635
670
FEEDBACK TRANSFER FUNCTION
550
SIGNAL PROCESSOR
630
REFERENCE SHAPING FILTER
730
740
640
ERROR SHAPING FILTER
ADAPTIVE FILTER
510
610
645
600
580
570
530
INPUT
520

Fig. 8



INPUT
120
INPUT
161
250
REFERENCE SHAPING FILTER
270
275
ADAPTIVE FILTER
110
290
280
+
-
140
100
295
ERROR SHAPING FILTER
310
350
SIGNAL PROCESSOR
300
D/A
305
307
308

Fig. 9

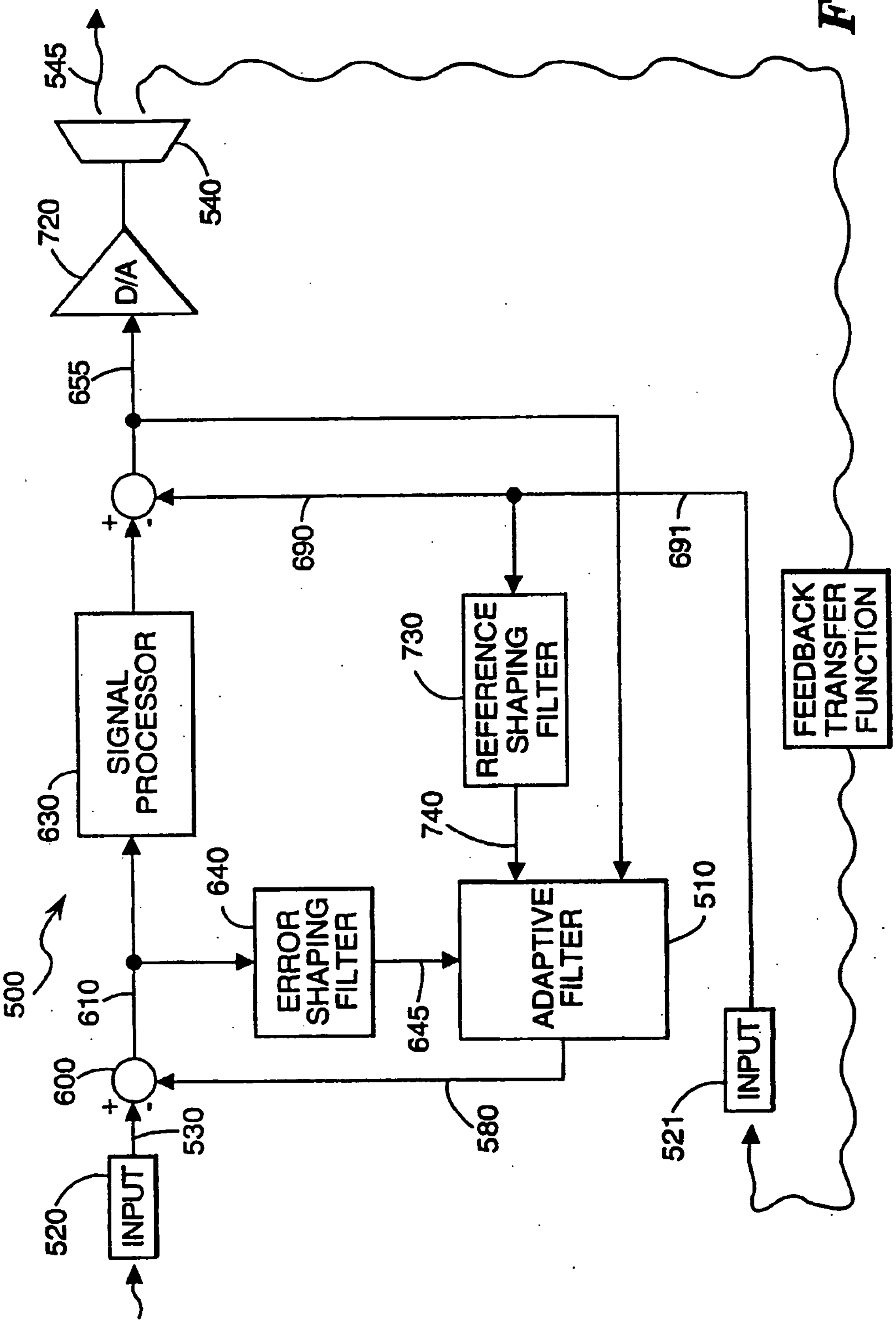

545
540
720
D/A
655
690
691
730
REFERENCE SHAPING FILTER
SIGNAL PROCESSOR
630
740
640
ERROR SHAPING FILTER
ADAPTIVE FILTER
510
500
610
645
FEEDBACK TRANSFER FUNCTION
600
530
INPUT
520
580
INPUT
521
+
-
+
-

Fig. 10

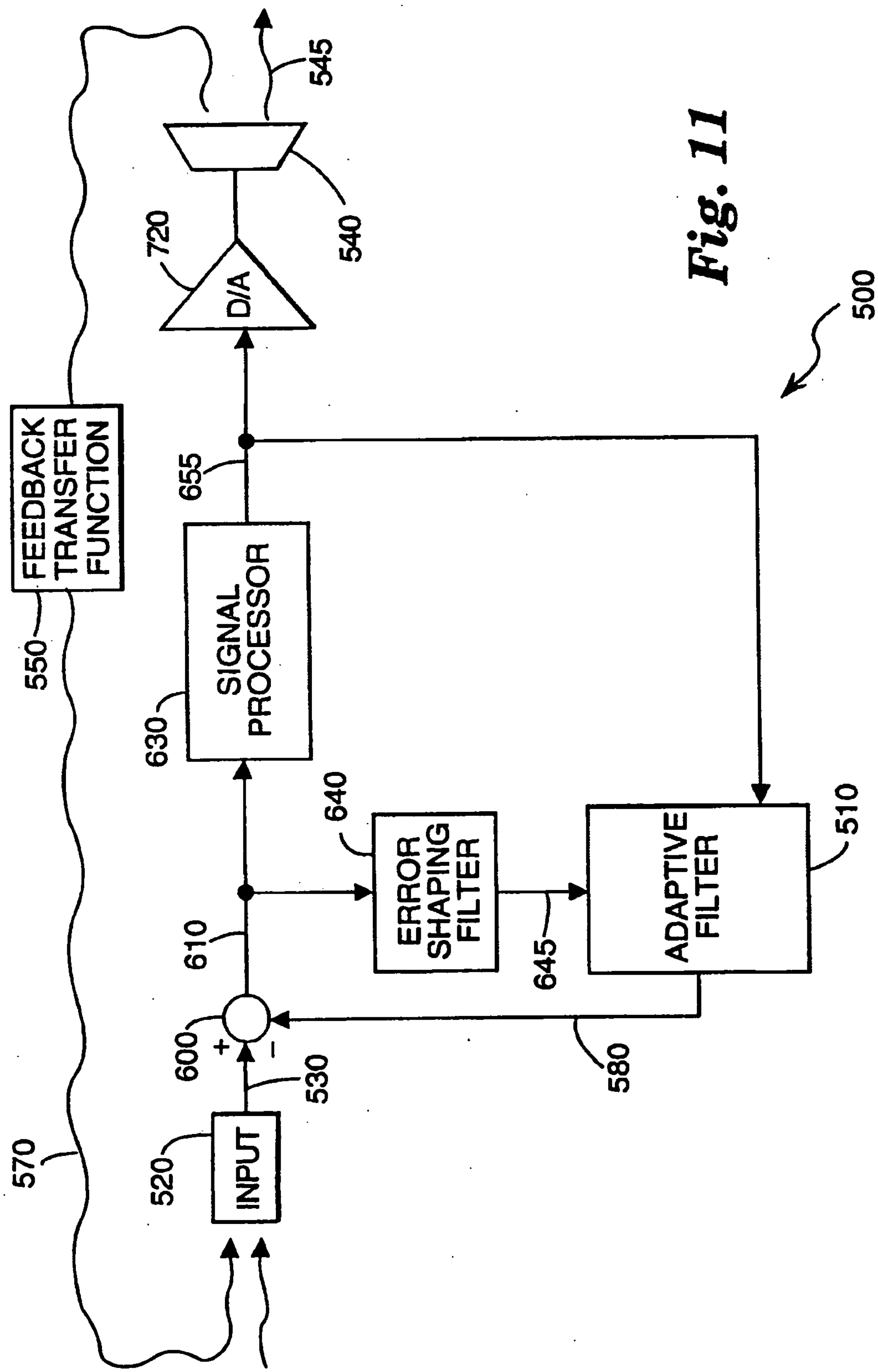

FEEDBACK TRANSFER FUNCTION
550
570
545
540
720
D/A
655
SIGNAL PROCESSOR
630
640
ERROR SHAPING FILTER
ADAPTIVE FILTER
510
500
610
645
600
+
-
530
580
520
INPUT

Fig. 11

AUDITORY PROSTHESIS, NOISE SUPPRESSION APPARATUS AND FEEDBACK SUPPRESSION APPARATUS HAVING FOCUSED ADAPTIVE FILTERING

The present invention relates generally to auditory prosthesis, noise suppression apparatus and feedback suppression apparatus used in acoustical systems, and particularly to such prostheses and apparatus having adaptive filtering.

BACKGROUND OF THE INVENTION

Designers of audio signal processing systems including auditory prostheses face the continuing challenge of attempting to eliminate feedback and noise from an input signal of interest. For example, a common complaint among users of auditory prosthesis such as hearing aids is their inability to understand speech in a noisy environment. In the past, hearing aid users were limited to listening-in-noise strategies such as adjusting the overall gain via volume control, adjusting the frequency response, or simply removing the hearing aid. More recent hearing aids have used noise reduction techniques based on, for example, the modification of the low frequency gain in response to noise. Typically, however, these strategies and techniques have been incapable of achieving a desired degree of noise reduction.

Many commercially available hearing aids are also subject to the distortion, ringing and squealing engendered by acoustical feedback. This feedback is caused by the return to the input microphone of a portion of the sound emitted by the acoustical hearing aid output transducer. Such acoustical feedback may propagate either through or around an earpiece used to support the transducer.

In addition to effectively reducing noise and feedback, a practical ear-level hearing aid design must accommodate the power, size and microphone placement limitations dictated by current commercial hearing aid designs. While powerful digital signal processing techniques are available, they require considerable space and power in the hearing aid hardware and processing time in the software. The miniature dimensions of hearing aids place relatively rigorous constraints on the space and power which may be devoted to noise and feedback suppression.

One approach to remedying the distortion precipitated by noise and feedback interference involves the use of adaptive filtering techniques. The frequency response of the adaptive filter can be made to self-adjust sufficiently rapidly to remove statistically "stationary" (i.e., slowly-changing) noise components from the input signal. Adaptive interference reduction circuitry operates to eliminate stationary noise across the entire frequency spectrum, with greater attenuation being accorded to the frequencies of high energy noise. However, environmental background noise tends to be concentrated in the lower frequencies, in most cases below 1,000 Hertz.

Similarly, undesirable feedback harmonics tend to build up in the 3,000 to 5,000 Hertz range where the gain in the feedback path of audio systems tends to be the largest. As the gain of the system is increased the distortion induced by feedback harmonics introduces a metallic tinge to the audible sound. Distortion is less pronounced at frequencies below 3,000 Hertz as a consequence of the relatively lower gain in the feedback path.

Although background noise and feedback energy are concentrated in specific spectral regions, adaptive noise filters generally operate over the entire bandwidth of the hearing aid. Adaptive noise filters typically calculate an estimate of noise by appropriately adjusting the weighting parameters of a digital filter in accordance with the Least Mean Square (LMS) algorithm, and then use the estimate to minimize noise. The relationship between the mean square error and the N weight values of the adaptive filter is quadratic. To minimize the mean square error, the weights are modified according to the negative gradient of an error surface obtained by plotting the mean square error against each of the N weights in N dimensions. Each weight is then updated by (i) computing an estimate of the gradient; (ii) scaling the estimate by a scaler adaptive learning constant, μ; and (iii) subtracting this quantity from the previous weight value.

This full-frequency mode of adjustment tends to skew the noise and feedback suppression capability of the filter towards the frequencies of higher signal energy, thereby minimizing the mean-square estimate of the energy through the adaptive filter. However, the set of parameters to which the adaptive filter converges when the full noise spectrum is evaluated results in less than desired attenuation over the frequency band of interest. Such "incomplete" convergence results in the noise and feedback suppression resources of the adaptive filter not being effectively concentrated over the spectral range of concern.

Accordingly, a need in the art exists for an adaptive filtering system wherein noise or feedback suppression capability is focused over a selected frequency band.

SUMMARY OF THE INVENTION

In summary, the present invention comprises a noise and feedback suppression apparatus for processing an audio input signal having both a desired component and an undesired component. When implemented so as to effect noise cancellation the present invention includes a first filter operatively coupled to the input signal. The first filter generates a reference signal by selectively passing an audio spectrum of the input signal which primarily contains the undesired component. The reference signal is supplied to an adaptive filter disposed to filter the input signal so as to provide an adaptive filter output signal. A combining network operatively coupled to the input signal and to the adaptive filter output signal uses the adaptive filter output signal to cancel the undesired component from the input signal and create an error signal. The noise suppression apparatus further includes a second filter for selectively passing to the adaptive filter an audio spectrum of the error signal substantially encompassing the spectrum of the undesired component of the input signal. This cancellation effectively removes the undesired component from the input signal without substantially affecting the desired component of said input signal.

When implemented to suppress feedback within, for example, a hearing aid, the present invention includes a combining network operatively coupled to an input signal and to an adaptive filter output signal. The combining network uses the adaptive-filter output signal to cancel the feedback component from the input signal and thereby deliver an error signal to a hearing aid signal processor. The inventive feedback suppression

circuit further includes an error filter disposed to selectively pass a feedback spectrum of the error signal to the adaptive filter. A reference filter supplies a reference signal to the adaptive filter by selectively passing the feedback spectrum of the noise signal, wherein the adaptive filter output signal is synthesized in response to the reference signal.

In a preferred embodiment, a noise probe signal is inserted into the output signal path of the feedback suppression circuit to supply a source of feedback during times of little containment of the undesired feedback signal being present within the audio environment of the circuit. The noise probe signal may also be supplied directly to the adaptive filter to aid in the convergence of the adaptive filter.

Optionally, a second microphone may be used in place of input delay of the noise suppression circuit or in place of the noise probe signal in the feedback suppression circuit.

BRIEF DESCRIPTION OF THE DRAWINGS

Additional objects and features of the invention will be more readily apparent from the following detailed description and appended claims when taken in conjunction with the drawings, in which:

FIG. 1 is a simplified block diagrammatic representation of a noise suppression apparatus of the present invention as it would be embodied in an auditory prosthesis;

FIG. 2 shows a detailed block diagrammatic representation of the noise suppression apparatus of the present invention;

FIG. 3 is a flow chart illustrating the manner in which successive input samples to the inventive noise suppression circuit are delayed by an J-sample delay line;

FIG. 4 depicts a flow chart outlining the manner in which an FIR implementation of a shaping filter processes a stream of delayed input samples produced by the J-sample delay line;

FIG. 5 is a flow chart illustrating the process by which an adaptive signal comprising a stream of samples y(n) is synthesized by an adaptive filter;

FIG. 6 is a block diagrammatic representation of an optional post-filter network coupled to the adaptive filter;

FIG. 7 depicts a top-level flow chart describing operation of the noise suppression apparatus of the present invention;

FIG. 8 is a block diagram depiction of the feedback suppression apparatus of the present invention as it would be embodied in an auditory prosthesis;

FIG. 9 is a block diagram of a two microphone implementation of the noise suppression apparatus of the present invention;

FIG. 10 is a block diagram of a two microphone implementation of the feedback suppression apparatus of the present invention; and

FIG. 11 is a block diagram of an alternative embodiment of the feedback suppression apparatus of the present invention.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

The noise suppression and feedback cancellation circuits of the present invention operate to focus the adaptive filtering systems included therein over particular frequency bands of interest. In this way adaptive filtering capacity is concentrated in a predefined manner, thereby enabling enhanced convergence of the adaptive filter across the noise and feedback bands of concern. The present invention focuses filtering resources in this manner by employing shaping filters disposed to selectively transmit energy from specific spectral bands to the adaptive filter included within each circuit.

Noise Suppression Circuit

Referring to FIG. 1, a noise suppression circuit **100** for use in auditory prosthesis such as hearing aids uses a time-domain method for focusing the bandwidth over which undesired noise energy is suppressed. As is described more fully below, the noise elimination band of an adaptive filter **110** is defined by selectively pre-filtering reference and error inputs provided to adaptive filter **110**. This signal shaping focuses noise suppression circuit **100** on the frequency band of interest, thus resulting in efficient utilization of the resources of adaptive filter **110**.

Noise suppression circuit **100** has an input **120** representative of any conventional source of a hearing aid input signal such as that produced by a microphone, signal processor, or the like. Input **120** also includes an analog to digital converter (not shown) for analog inputs so that the input signal **140** is a digital signal. Input signal **140** is received by an J-sample delay **160** and by a signal combiner **280**. Delay **160** serves to decorrelate, in time, delayed input signal **250** supplied to adaptive filter **110** from input signal **140**. The length of delay **160** will generally be selected to be of a duration which preserves the auto-correlation between noise energy within input signal **140** and delayed input signal **250** yet which significantly reduces the auto-correlation of the speech energy within the two signals. Specifically, delay **160** will preferably be sufficiently long to reduce the auto-correlation of the speech energy within input signal **140** and delayed input signal **250** such that minimum speech cancellation occurs through the adaptive filtering process. For example, at a 10 kiloHertz sampling rate, an eight sample delay results in an acceptable time delay of eight hundred microseconds. It is also believed that such a delay will preserve the auto-correlation between the noise energy within input signal **140** and delayed input signal **250** to the extent required to enable a suitable degree of noise cancellation.

In an alternative implementation of the inventive noise suppression circuit illustrated in FIG. 9, a second microphone **161** is used instead of delay circuit **160** to provide the reference signal **250**. Second microphone **161** will preferably be positioned so as to receive primarily only ambient noise energy and a minimum of audible speech. In this way the sampled version of the electrical signal generated by second microphone **161** will be substantially uncorrelated with the speech information inherent within sampled input signal **140**, thus preventing significant speech cancellation from occurring during adaptive filtering. Microphone **120** and second microphone **161** will, however, typically be located within the same noise field such that at least some degree of correlation exists between noise energy within input signal **140** and reference signal **250** provided by second microphone **161**.

Continuing in the description of FIGS. 1 and 9, delayed (with respect to FIG. 1) input signal **250** is also transmitted to reference shaping filter **270** disposed to provide focused reference signal **275** to adaptive filter **110**. Reference shaping filter **270** is preferably realized

as a finite impulse response (FIR) filter having a transfer characteristic which passes a noise spectrum desired to be removed from input signal **140**, but does not pass most of the speech spectrum of interest. Noise from machinery and other distracting background noise is frequently concentrated at frequencies of less than one hundred Hertz, while the bulk of speech energy is present at higher audible frequencies. Accordingly, reference shaping filter **270** will preferably be of a low-pass variety having a cut-off frequency of less than, for example, several hundred Hertz. When an FIR implementation is employed, the tap weights included within reference shaping filter **270** may be determined from well-known FIR filter design techniques upon specification of the desired low-pass cut-off frequency. See, for example, U.S. Pat. No. 4,658,426, Chabries et al, Adaptive Noise Suppressor, the contents of which are hereby incorporated by reference.

Referring again to FIG. **1**, an adapted signal **290** synthesized by adaptive filter **110** is supplied to signal combiner **280**. Adapted signal **290**, which characterizes the noise component of the input signal **140**, is subtracted from input signal **140** by combiner **280** in order to provide a desired output signal **295** to signal processor **300**. Signal processor **300** preferably includes a filtered amplifier circuit designed to increase the signal energy over a predetermined band of audio frequencies. In particular, signal processor **300** may be realized by one or more of the commonly available signal processing circuits available for processing digital signals in hearing aids, For example, signal processor **300** may include the filter-limit-filter structure disclosed in U.S. Pat. No. 4,548,082, Engebretson et al, the contents of which are hereby incorporated by reference. After desired output signal **295** has passed through signal processor **300**, a digital to analog converter **305** converts resulting signal **302** into analog signal **307**. Analog signal **307** drives output transducer **308** disposed to generate an acoustical waveform in response thereto.

Desired output signal **295** is also provided to error shaping filter **310** having a passband chosen to transmit the spectral noise range desired to be eliminated from input signal **140**. Error shaping filter **310** is preferably a finite impulse response (FIR) filter having a transfer characteristic which passes a noise spectrum desired to be removed from input signal **140**, but does not pass most of the speech spectrum of interest. Hence, error shaping filter **310** will preferably be of a low-pass variety having a cut-off frequency substantially identical to that to reference shaping filter **270** (i.e., of less than several hundred Hertz).

The noise suppression circuit **100** is depicted in greater detail within the block diagrammatic representation of FIG. **2**. Referring to FIG. **2**, samples x(n) of input signal **140** are initially delayed by processing the signals through J-sample delay **160**. The samples of delayed input signal **250**, denoted by x(n-J), are then further processed by reference shaping filter **270**. As is described more fully below, the resultant stream of samples $U_w(n)$ of focused reference signal **275** along with the weighted error signal $e_w(n)$ of filtered error stream **350** computed during the preceding cycle of adaptive filter **110** are used to update tap weights h(n) within adaptive filter **110**.

Subsequent to modification of the adaptive weights h(n), adaptive filter **110** processes samples x(n-J) in order to generate adaptive signal **290**. In this way, adapted signal **290** is made available to combiner **280**, which produces desired output signal **295** by subtracting samples of adapted signal **290** from samples x(n) of input signal **140**. Desired output signal **295** is then supplied to error shaping filter **310** to allow computation of the samples $e_w(n)$ of filtered error stream **350** to be used during the next processing cycle of adaptive filter **110**.

The operation of noise suppression circuit **100** may be more specifically described with reference to the signal flow charts of FIGS. **3**, **4**, **5** and **6**. In particular, the flow chart of FIG. **3** illustrates the manner in which successive samples of input signal **140** are delayed by J-sample delay **160**. J-sample delay **160** is preferably implemented as a serial shift register, receiving samples from input signal **140** and outputting each received sample after J sample periods. As is indicated in FIG. **3**, during each sampling period the "oldest" sample x(J) included in the shift register becomes the current sample of delayed input signal **250**. The remaining values x(i) are then shifted one tap in the filter. The current sample of input signal **140** is stored as value x(1).

FIG. **4** depicts a flow chart outlining the manner in which an FIR implementation of reference shaping filter **270** processes the stream of samples of delayed input signal **250** using a series of tap positions. Referring to FIG. **4**, during each sampling period, a first processing cycle is used to shift the existing data y(i) in reference shaping filter **270** by one tap position. As is typically the case, adjacent tap positions of reference shaping filter **270** are separated by single-unit delays (represented by the notation "z^{-1}" in FIG. **2**). The current sample of delayed input signal **250** is placed in the first tap location y(1) of reference shaping filter **270**. This first processing cycle is essentially identical to the update procedure for J-sample delay circuit **160** described above with reference to FIG. **3**.

Referring to FIGS. **2** and **4**, during a second cycle within the sample period, each filter sample y(i) is multiplied by a fixed tap weight a(i) having a value determined in accordance with conventional FIR filter design techniques. The sum of the tap weight multiplications is accumulated by M-input summer **340**, which provides focused reference signal **275** supplied to adaptive filter **110**.

FIG. **5** is a flow chart illustrating the process by which the stream of samples y(n) (defined earlier with respect to FIG. **2**) is synthesized by adaptive filter **110**. During a first cycle **342** within each sample period the current sample of focused reference signal **275** is shifted into adaptive filter **110** as adaptive input sample $u_w(1)$, wherein the subscript w signifies the "spectrally weighted" shaping effected by reference shaping filter **270**. The preceding N-1 reference samples are denoted as $u_w(2), u_w(3), \ldots u_w(N)$, and are each shifted one tap location within adaptive filter **110** as the sample $u_w(1)$ is shifted in. Once this alignment process has occurred, a second cycle **344** is initiated wherein adaptive weights $h(1), h(2), \ldots h(N)$ are modified in accordance with the current value e_w of the filtered error stream **350**. As is explained more fully below, this updating process is carried out in accordance with the following recursion formula:

$$h(i)_{NEW} = h(i)_{OLD}(1-\beta) + \mu u_w(i) e_w \qquad \text{(Equation 1)}$$

where (i) represents the i^{th} component of adaptive filter **110**, μ is an adaption constant determinative of the rate of convergence of adaptive filter **110**, and β is a real number between zero and one. The value of μ will

preferably be chosen in the conventional manner such that adaptive filter 110 converges at an acceptable rate, but does not become overly sensitive to minor variations in the power spectra of input signal 140.

In a third cycle 346, the delayed samples x(n-J-i+1) in the N-tap delay line of adaptive filter 110 are shifted by one tap position, and in a fourth cycle 348 the updated adaptive filter weights h(i) are multiplied by the delayed samples x(n-J-i+1) and summed to generate the current sample of adapted signal 290 as output from adaptive filter 110. The index "n-J-i+1" for the delayed samples indicates the J sample period delay associated with J-sample delay 160, plus the delay associated with adaptive filter 110.

Equation (1) above is based on a "leaky least means square" error minimization algorithm commonly understood by those skilled in the art and more fully described in Haykin, *Adaptive Filter Theory*, Prentice-Hall (1986), p. 261, which is incorporated herein by reference. This choice of adjustment algorithm allows that, in the absence of input, the filter coefficients of adaptive filter 110 will adjust to zero. In this way adaptive filter 110 is prevented from self-adjusting to remove components from input signal 140 not included within the passband of reference shaping filter 270 and error shaping filter 310. Those skilled in the art will recognize that other adaptive filters and algorithms could be used within the scope of the invention. For example, a conventional least means square (LMS) algorithm such as is described in Widrow, et al., *Adaptive Noise Canceling: Principles and Applications*, Proceedings of the IEEE, 63(12), 1692-1716 (1975), which is incorporated herein by reference, may be employed in conjunction with a low-pass post-filter network 380 shown in FIG. 6. The filter network 380 serves to minimize the possibility that filtering characteristics will be developed based on information included within the frequency spectrum outside of the passband of reference shaping filter 270 and error shaping filter 310.

As is indicated by FIG. 6, the filter network 380 includes a low-pass filter 390 addressed by adaptive signal 290. Low pass filter 390 preferably has a low-pass transfer characteristic and, preferably is substantially similar to those of reference shaping filter 270 and error shaping filter 310. Filter network 380 further includes a K-sample delay 410 coupled to input signal 140 for providing a delay equivalent to that of low pass filter 390. Summation node 420 subtracts the output of low pass filter 390 from that of K-sample delay 410 and provides the difference to signal processor 300.

In conventional adaptive filtering schemes implementing some form of the LMS algorithm, the coefficients of the adaptive filter are updated to minimize the expected value of the squared difference between input and reference signals over the entire system bandwidth. In contrast, reference shaping filter 270 and error shaping filter 310 of the present invention focus adaptive cancellation over a desired spectral range. Specifically, reference shaping filter 270 and error shaping filter 310 are M^{th}-order FIR spectral shaping filters and may be represented by coefficient vector W:

$$W=[w(1), w(2), \ldots w(M)]^T, \qquad \text{(Equation 2)}$$

where T denotes the vector transpose. The difference between the stream of samples x(n) from input signal 140 and the stream of samples y(n) from adapted signal 290 may be represented by error vector E(n), in which

$$E(n)=[e(n), e(n-1), \ldots e(n-M+1)]^T \qquad \text{(Equation 3)}$$

which represents the set of error values stored in delay line 420 of error shaping filter 310. Filtered error stream 350 (FIG. 2) is spectrally weighted and the expected mean-square of which it is desired to minimize, is given by

$$e_w(n)=[W]^T \cdot E(n). \qquad \text{(Equation 4)}$$

The coefficient vector H=[h(1), h(2), ... h(N)] of the adaptive filter 110 which minimizes the expectation of the square of Equation 4 may be represented as

$$H=E\{[U_w(n)\cdot[U_w(n)]^T]^{-1}\}\cdot E\{x_w(n)\cdot U_w(n)\} \qquad \text{(Equation 5)}$$

where $x_w(n)$ is a weighted sum of the samples of input signal 140, defined as

$$x_w(n)=[W]^T \cdot X(n), \qquad \text{(Equation 6)}$$

where

$$X(n)=[x(n), x(n-1), \ldots x(n-M+1)]^T. \qquad \text{(Equation 7)}$$

In Equation 5, $U_w(n)$ denotes the vector of the spectrally weighted samples of focused reference signal 275, where

$$U_w(n)=[u_w(n), u_w(n-1), \ldots u_w(n-N+1)]^T, \text{ and} \qquad \text{(Equation 8)}$$

$$u_w(n)=[W]^T \cdot U(n), \qquad \text{(Equation 9)}$$

in which U(n) represents the stream of samples from delayed input signal 250.

Equations 2 through 9 describe the parameters included within the spectrally weighted LMS update algorithm of Equation 1 (see above). The adaptive weights h(i) of adaptive filter 110 are modified each sample period by the factor B, wherein $B=1-\beta$, via scaling blocks 450 (FIG. 2) in order to implement the "leaky" LMS algorithm given by Equation 1.

It is noted that the primary signal processing path, which includes input 120 as well as signal processor 300 and output transducer 308, is uninterrupted except for the presence of signal combiner 280. That is, the reference and error time sequences to adaptive filter 110 are shaped without corrupting the primary signal path with the finite precision weighting filters typically required in the implementation of conventional frequency-weighted noise-cancellation approaches.

FIG. 7 depicts a top-level flow chart describing operation of noise suppression circuit 100. In the following discussion the term "execute" implies that one of the operative sequences described with reference to FIGS. 3, 4 and 5 is performed in order to accomplish the indicated function. Referring to FIGS. 2 and 7, the current sample of input signal 140 is initially delayed (1710) by processing the signal through J-sample delay 160. The samples of delayed input signal 250 are then further processed (1720) by reference shaping filter 270. The resultant stream of samples of focused reference signal 275 along with the weighted error signal of filtered error stream 350 computed during the preceding cycle of adaptive filter 110 enable execution of the adaptive weight update routine (1730).

As is indicated by FIG. 7, subsequent to modification of the adaptive weights, adaptive filter 110 processes (1740) delayed input signal 250 in order to generate adaptive signal 290. In this way, adapted signal 290 is made available to combiner 280, which produces desired output signal 295 by subtracting (1750) adapted signal 290 from input signal 140. Desired output signal 295 is then supplied to error shaping filter 310 to allow computation (1760) of filtered error stream 350 to be used during the next processing cycle of adaptive filter 110. The process described with reference to FIG. 7 occurs during each sample period, at which time a new sample of input signal 140 is provided by input 120 and a new desired output signal 295 is supplied to signal processor 300.

Feedback Suppression Circuit

FIG. 8 shows a feedback suppression circuit 500 in accordance with the present invention, adapted for use in a hearing aid (not shown). Feedback suppression circuit 500 uses a time-domain method for substantially canceling the contribution made by undesired feedback energy to incident audio input signals. As is described more fully below, the feedback suppression band of adaptive filter 510 included within feedback suppression circuit 500 is defined by selectively pre-filtering filtered reference noise signal 740 and filtered error signal 645 provided to adaptive filter 510. This signal shaping focuses the circuit's feedback cancellation capability on the frequency band of interest (e.g. 3 to 5 kiloHertz), thus resulting in efficient utilization of the resources of adaptive filter 510. In this way, the principles underlying operation of feedback suppression circuit 500 are seen to be substantially similar to those incorporated within noise suppression circuit 100 shown in FIG. 1, with specific implementations of each circuit being disposed to reduce undesired signal energy over different frequency bands.

Referring to FIG. 8, feedback suppression circuit 500 has an input 520 which may be any conventional source of an input signal including, for example, a microphone and signal processor. A microphone (not shown) preferably included within input 520 generates an electrical input signal 530 from sounds external to the user of the hearing aid, from which is synthesized an output signal used by output transducer 540 to emit filtered and amplified sound 545. Input 520 also includes an analog to digital converter (not shown) so that input signal 530 is a digital signal. As is indicated by FIG. 8, some of the sound 545 emitted by output transducer 540 returns to the microphone within input 520 through various feedback paths generally characterized by feedback transfer function 550. Feedback signal 570 is a composite representation of the aggregate acoustical feedback energy received by input 520.

Adaptive output signal 580 generated by adaptive filter 510 is subtracted from input signal 530 by input signal combiner 600 in order to produce a feedback canceled signal 610. Feedback canceled signal 610 is supplied both to signal processor 630 and to error shaping filter 640. Signal processor 630 preferably is implemented in the manner described above with reference to signal processor 300 of noise cancellation circuit 100. Output 635 of signal processor 630 is added at summation node 650 to broadband noise signal 690 generated by noise probe 670. Composite output signal 655 created at summation node 650 is provided to digital-to-analog converter 720 and adaptive filter 510. The output of digital-to-analog converter 720 is submitted to output transducer 540.

Noise probe 690 also supplies noise reference input 691 to reference shaping filter 730 which in turn is coupled to adaptive filter 510. Broadband noise signal 690 and noise reference signal 691 generated by noise probe 670 are preferably identical, and ensure that adaptive operation of feedback cancellation circuit 500 is sustained during periods of silence or minimal acoustical input. Specifically, the magnitude of broadband noise signal 690 provided to summation node 650 should be large enough to ensure that at least some acoustical energy is received by input 520 (as a feedback signal 570) in the absence of other signal input. In this way, the weighting coefficients within adaptive filter 510 are prevented from "floating" (i.e. from becoming randomly arranged) during periods of minimal audio input. Noise probe 670 may be conventionally realized with, for example, a random number generator operative to provide a random sequence corresponding to a substantially uniform, wideband noise signal. The broadband noise signal 690 can be provided at a level below the auditory threshold of users, usually significantly hearing-impaired users, and is perceived as a low-level white noise sound by those afflicted with less severe hearing losses.

When noise probe 670 is operated, a faster convergence of adaptive filter 510 generally can be obtained by breaking the main signal path by temporarily disconnecting the output of signal processor 630 from combiner 650.

Alternatively as shown in FIG. 10, second microphone 521 may be used in lieu of the noise probe 670 to provide the reference signals 690 and 691. As was discussed with reference to FIG. 9, such second microphone 521 will preferably be positioned a sufficient far from the microphone preferably included within input 520 to prevent cancellation of speech energy within input signal 530.

Continuing with reference to FIGS. 8 and 10, filtered reference noise signal 740 applied to modify the weights of adaptive filter 510 is created by passing noise reference signal 691 through reference shaping filter 730. Error shaping filter 640 and reference shaping filter 730 preferably will be realized as finite impulse response (FIR) filters governed by a transfer characteristic formulated to pass a feedback spectrum (e.g., 3 to 5 kiloHertz) desired to be removed from input signal 530. Because the speech component of input signal 530 is not present within reference noise signal 691, the speech energy within input signal 530 will be uncorrelated with adaptive output signal 580 synthesized by adaptive filter 510 from noise reference signal 691. As a consequence, the speech component of input signal 530 is left basically intact subsequent to combination with adaptive output signal 580 at signal combiner 600 irrespective of the extent to which shaping filters (640 and 730) transmit signal energy within the frequency realm of intelligent speech. This enables the transfer characteristics of the shaping filters (640 and 730) to be selected in an unconstrained manner to focus the feedback cancellation resources of the feedback suppression circuit 500 over the spectral range in which the gain in feedback transfer function 550 is the largest.

Determination of feedback transfer function 550 may be accomplished empirically by transmitting noise energy from the location of output transducer 540 and

measuring the acoustical waveform of feedback signal 570 received at input 520.

Alternatively, feedback transfer function 550 may be analytically estimated when particularized knowledge is available with regard to the acoustical characteristics of the environment between output transducer 540 and input 520. For example, information relating to the acoustical properties of the human ear canal and to the specific physical structure of the hearing aid could be utilized to analytically determine feedback transfer function 550.

FIG. 11 illustrates an alternative embodiment of the feedback suppression apparatus of the present invention. Since the feedback suppression apparatus previously illustrated in FIG. 8 typically may be used in environments having a level of noise, it is possible in some circumstances to eliminate the noise probe generator 670 of FIG. 8. As illustrated in FIG. 11, eliminating the noise probe generator enables adaptive filter 510 to rely of presence of some noise in the output 655 of signal processor 630 in frequency band of interest. Adaptive filter 510 adapts only to error shaping filter 640, which focuses the adaptive energy of adaptive filter 510 to the portion of incoming signal containing the feedback component, and to signal 655 output from signal processor 630. Output 655 of signal processor 630 is fed directly to the input of adaptive filter 510 and to digital-to-analog converter 720.

While the present invention has been described with reference to a few specific embodiments, the description is illustrative of the invention and is not to be construed as limiting the invention. Various modifications may occur to those skilled in the art without departing from the true spirit and scope of the invention as defined by the appended claims. For example, algorithms other than the LMS filter algorithm may be used to control the adaptive filters included within noise suppression circuit 100 and feedback cancellation circuit 500. Similarly, shaping filters (270, 310, 640 and 730) may be tuned so as to focus adaptive filtering to eliminate undesired signal energy over spectral ranges other than those disclosed herein.

What is claimed is:

1. A noise suppression apparatus for processing an audio input signal having both a desired component and an undesired component, comprising:

first filter means operatively coupled to said input signal for generating a reference signal by selectively passing an audio spectrum of said input signal containing primarily said undesired component;

adaptive filter means operatively coupled to said input signal and to said reference signal for adaptively filtering said input signal in order to provide an adaptive filter output signal;

combining means operatively coupled to said input signal and to said adaptive filter output signal for combining said adaptive filter output signal with said input signal to cancel said undesired component from said input signal and produce an error signal; and

second filter means receiving said error signal for selectively passing to said adaptive filter means an audio spectrum of said error signal corresponding to said undesired component of said input signal;

said adaptive filter means being controlled in accordance with a signal filtering algorithm that employs both said input signal selectively passed by said first filter and said selectively passed error signal;

whereby said undesired component is effectively removed from said input signal without substantially affecting said desired component of said input signal.

2. The apparatus of claim 1 further including decorrelation means inserted between said input signal and said first filter means, and between said input signal and said adaptive filter means, for decorrelating said input signal from said adaptive filter output signal.

3. The apparatus of claim 2 wherein said decorrelation means comprises a signal delay circuit that delays transmission of said input signal.

4. The apparatus of claim 3 wherein said input signal comprises a digital signal obtained by sampling an analog signal during successive sample periods, and wherein said signal delay circuit delays transmission of said digital signal by at least four of said sample periods.

5. The apparatus of claim 1 wherein said adaptive filter means is a FIR filter having a set of filter coefficients and means for periodically updating said filter coefficients, in accordance with values of said reference signal and a portion of said error signal passed by said second filter means, so as to minimize a predefined least means square error value.

6. The apparatus of claim 5 wherein said adaptive filter means further includes a low-pass post-filter network, said post-filter network including:

means for delaying said input signal,

a low-pass filter addressed by said adaptive filter output signal, and

a difference node operatively coupled to'said delayed input signal and to an output of said low-pass filter.

7. The apparatus of claim 1 wherein said adaptive filter means is a FIR filter having filter coefficients h(i) and coefficient updating means for updating said filter coefficients in accordance with a leaky least means square update function of the form:

$$h_{new}(i)=(1-\beta)h_{old}(i)+\mu u_2(i)e_w$$

wherein μ is an adaptation constant, β is a real number between zero and one, $h_{new}(i)$ represents an i^{th} filter coefficient's updated value, $h_{old}(i)$ represents said i^{th} filter coefficient's previous value, $u_w(i)$ denotes an i^{th} sample of the reference signal, and e_w denotes the portion of said error signal passed by said second filter means.

8. The apparatus of claim 1 wherein spectral energy included within said undesired component, within said reference signal, and within said filtered error signal is generally confined to frequencies below 1 kiloHertz.

9. For use in an audio system having microphone means for generating an input signal from sounds external to said system and transducer means for emitting sound in response to an output signal provided by signal processing means, wherein a portion of the sound emitted by said transducer means propagates to the microphone means to add a feedback signal to the input signal, a feedback suppression apparatus comprising:

probe means for generating a noise signal, said noise signal being injected into said output signal;

combining means operatively coupled to said input signal and to an adaptive filter output signal for subtracting said adaptive filter output signal from said input signal so as to substantially cancel said feedback signal from said input signal and to gener-

ate an error signal that is input into said signal processing means;

first filter means operatively coupled to said error signal for generating a filtered error signal by selectively passing an audio spectrum of said error signal corresponding to said feedback signal's audio spectrum;

adaptive filter means operatively coupled to said filtered error signal for generating said adaptive filter output signal and for providing said adaptive filter output signal to said combining means; and

second filter means for selectively passing to said adaptive filter means an audio spectrum of said noise signal corresponding to said feedback signal's audio spectrum.

10. The apparatus of claim 9 wherein said first and second filter means respectively include first and second FIR filters having passbands encompassing the spectral range between 3 and 5 kiloHertz.

11. The apparatus of claim 9 wherein said adaptive filter means is a FIR filter having a set of filter coefficients and including means for periodically updating said filter coefficients, in accordance with values of said filtered error signal and a portion of said noise signal passed by said second filter means, so as to minimize a predefined least means square error value.

12. The apparatus of claim 9 wherein said adaptive filter means is a FIR filter having filter coefficients h(i) and coefficient updating means for updating said filter coefficients in accordance with a leaky least means square update function of the form:

$$h_{new}(i)=(1-\beta)h_{old}(i)+\mu u_w(i)e_w$$

wherein μ is an adaptation constant, β is a real number between zero and one, $h_{new}(i)$ represents an i^{th} filter coefficient's updated value, $h_{old}(i)$ represents said i^{th} filter coefficient's previous value, $u_w(i)$ denotes an i^{th} sample of the reference signal, and e_w denotes the portion of said error signal passed by said second filter means.

13. The apparatus of claim 9 wherein spectral energy included within said filtered error signal is generally confined to frequencies between 3 and 5 kiloHertz.

14. The apparatus of claim 9 wherein said probe means includes a random number generator for introducing a sequence of random numbers into said noise signal.

15. An auditory prosthesis disposed to process acoustical signal energy, comprising:

a microphone for generating an audio input signal in response to said acoustical signal energy, said input signal having both a desired component and an undesired component;

first filter means operatively coupled to said input signal for generating a reference signal by selectively passing an audio spectrum of said input signal containing primarily said undesired component;

adaptive filter means operatively coupled to said input signal and to said reference signal for adaptively filtering said input signal in order to provide an adaptive filter output signal;

combining means operatively coupled to said input signal and to said adaptive filter output signal for combining said adaptive filter output signal with said input signal to cancel said undesired component from said input signal and produce an error signal;

second filter means operatively coupled to said error signal for selectively passing to said adaptive filter means an audio spectrum of said error signal corresponding to said undesired component of said input signal;

said adaptive filter means being controlled in accordance with a signal filter algorithm that employs both said reference signal and a portion of said error signal passed by said second filter means;

a signal processor having an input coupled to said error signal and producing an desired output signal;

output transducer means for emitting sound in response to said desired output signal;

whereby said undesired component is effectively removed from said input signal without substantially affecting said desired component of said input signal.

16. The auditory prosthesis of claim 15 further including decorrelation means inserted between said input signal and said first filter means, and between said input signal and said adaptive filter means, for decorrelating said input signal from said adaptive filter output signal.

17. The auditory prosthesis of claim 16 wherein said decorrelation means comprises a signal delay circuit that delays transmission of said input signal.

18. The auditory prosthesis of claim 17 wherein said input signal comprises a digital signal obtained by sampling an analog signal during successive sample periods, and wherein said signal delay circuit delays transmission of said digital signal by at least four of said sample periods.

19. The auditory prosthesis of claim 15 wherein said adaptive filter means is a FIR filter having a set of filter coefficients and including means for periodically updating said filter coefficients, in accordance with values of said reference signal and a portion of said error signal passed by said second filter means, so as to minimize a predefined least means square error value.

20. The auditory prosthesis of claim 19 wherein said adaptive filter means further includes a low-pass post-filter network, said post-filter network including:

means for delaying said input signal,

a low-pass filter addressed by said adaptive filter output signal, and

a difference node operatively coupled to said delayed input signal and to an output of said low-pass filter.

21. The auditory prosthesis of claim 15 wherein said adaptive filter means is a FIR filter having filter coefficients h(i) and coefficient updating means for updating said filter coefficients in accordance with a leaky least means square update function of the form:

$$h_{new}(i)=(1-\beta)h_{old}(i)+\mu u_w(i)e_w$$

wherein μ is an adaptation constant, β is a real number between zero and one, $h_{new}(i)$ represents an i^{th} filter coefficient's updated value, $h_{old}(i)$ represents said i^{th} filter coefficient's previous value, $u_w(i)$ denotes an i^{th} sample of the reference signal, and e_w denotes the portion of said error signal passed by said second filter means.

22. The auditory prosthesis of claim 15 wherein spectral energy included within said undesired component, within said reference signal, and within said filtered error signal is generally confined to frequencies below 1 kiloHertz.

23. An auditory prosthesis comprising:

microphone means for generating an input signal from sounds external to said prosthesis;

transducer means for emitting sound in response to an output signal, wherein a portion of the sound emitted by said transducer means propagates to the microphone means to add a feedback signal to the input signal;

signal processing means for producing said output signal;

probe means for generating a noise signal, said noise signal being injected into said output signal;

combining means operatively coupled to said input signal and to an adaptive filter output signal for subtracting said adaptive filter output signal from said input signal so as to substantially cancel said feedback signal from said input signal and to generate an error signal that is input into said signal processing means;

first filter means operatively coupled to said error signal for generating a filtered error signal by selectively passing an audio spectrum of said error signal corresponding to said feedback signal's audio spectrum;

second filter means for selectively passing an audio spectrum of said noise signal corresponding to said feedback signal's audio spectrum; and

adaptive filter means operatively coupled to said audio spectrum of said noise signal from said second filter means and to said filtered error signal for generating said adaptive filter output signal and for providing said adaptive filter output signal to said combining means.

24. The auditory prosthesis of claim **23** wherein said first and second filter means respectively include first and second FIR filters having passbands encompassing the spectral range between 3 and 5 kiloHertz.

25. The auditory prosthesis of claim **23** wherein said adaptive filter means is a FIR filter having a set of filter coefficients and means for periodically updating said filter coefficients, in accordance with values of said filtered error signal and a portion of said noise signal passed by said second filter means, so as to minimize a predefined least means square error value.

26. The auditory prosthesis of claim **23** wherein said adaptive filter means is a FIR filter having filter coefficients h(i) and coefficient updating means for updating said filter coefficients in accordance with a leaky least means square update function of the form:

$$h_{new}(i)=(1-\beta)h_{old}(i)+\mu u_w(i)e_w$$

wherein μ is an adaptation constant, β is a real number between zero and one, $h_{new}(i)$ represents an i^{th} filter coefficient's updated value, $h_{old}(i)$ represents said i^{th} filter coefficient's previous value, $u_w(i)$ denotes an i^{th} sample of the filtered error signal, and e_w denotes the portion of said error signal passed by said second filter means.

27. The auditory prosthesis of claim **23** wherein spectral energy included within said feedback component and within said filtered error signal is generally confined to frequencies between 3 and 5 kiloHertz.

28. The auditory prosthesis of claim **23** wherein said probe means includes a random number generator for introducing a sequence of random numbers into said noise signal.

29. For use in an audio system having input microphone means for generating an input signal from sounds external to said system and transducer means for emitting sound in response to an output signal provided by signal processing means, wherein a portion of the sound emitted by said transducer means propagates to the input microphone means to add a feedback signal to the input signal, a feedback suppression apparatus comprising:

reference microphone means responsive to said feedback signal for generating a noise signal, said noise signal being injected into said output signal;

combining means operatively coupled to said input signal and to an adaptive filter output signal for subtracting said adaptive filter output signal from said input signal so as to substantially cancel said feedback signal from said input signal and to generate an error signal that is input into said signal processing means;

first filter means operatively coupled to said error signal for generating a filtered error signal by selectively passing an audio spectrum of said error signal corresponding to said feedback signal's audio spectrum;

second filter means for selectively passing an audio spectrum of said noise signal corresponding to said feedback signal's audio spectrum; and

adaptive filter means operatively coupled to said audio spectrum of said noise signal and to said filtered error signal for generating said adaptive filter output signal and for providing said adaptive filter output signal to said combining means.

30. For use in an audio system having microphone means for generating an input signal from sounds external to said system and transducer means for emitting sound in response to an output signal provided by signal processing means, wherein a portion of the sound emitted by said transducer means propagates to the microphone means to add a feedback signal to the input signal, a feedback suppression apparatus comprising:

combining means operatively coupled to said input signal and to an adaptive filter output signal for subtracting said adaptive filter output signal from said input signal so as to substantially cancel said feedback signal from said input signal and to generate an error signal that is input into said signal processing means;

filter means operatively coupled to said error signal for generating a filtered error signal by selectively passing an audio spectrum of said error signal corresponding to said feedback signal's audio spectrum;

adaptive filter means operatively coupled to said filtered error signal for generating said adaptive filter output signal and for providing said adaptive filter output signal to said combining means.

31. The apparatus of claim **30** wherein said filter means comprise an FIR filter having a passband encompassing the spectral range between 3 and 5 kiloHertz.

32. The apparatus of claim **30** wherein said adaptive filter means is a FIR filter having a set of filter coefficients and including means for periodically updating said filter coefficients, in accordance with values of said filtered error signal and a portion of said error signal passed by said filter means, so as to minimize a predefined least means square error value.

33. The apparatus of claim **30** wherein said adaptive filter means is a FIR filter having filter coefficients h(i) and coefficient updating means for updating said filter

coefficients in accordance with a leaky least means square update function of the form:

$$h_{new}(i)=(1-\beta)h_{old}(i)+\mu u_w(i)e_w$$

wherein μ is an adaptation constant, β is a real number between zero and one, $h_{new}(i)$ represents an i^{th} filter coefficient's updated value, $h_{old}(i)$ represents said i^{th} filter coefficient's previous value, $u_w(i)$ denotes an i^{th} sample of the filtered error signal, and e_w denotes the portion of said error signal passed by said filter means.

34. The apparatus of claim 30 wherein spectral energy included within said feedback signal and within said filtered error signal is confined to frequencies between 3 and 5 kiloHertz.

* * * * *

UNITED STATES PATENT AND TRADEMARK OFFICE

CERTIFICATE OF CORRECTION

PATENT NO. : 5,402,496

DATED : March 28, 1995

INVENTOR(S) : Soli et al.

It is certified that error appears in the above-indentified patent and that said Letters Patent is hereby corrected as shown below:

Column 12, line 41, delete "$h_{new}(i) = (1-\beta)h_{old}(i) + \mu u_2(i)e_w$" and insert therefor -- $h_{new}(i) = (1-\beta)h_{old}(i) + \mu u_w(i)e_w$ --.

Signed and Sealed this

Twenty-second Day of August, 1995

Attest:

Bruce Lehman

BRUCE LEHMAN

Attesting Officer

Commissioner of Patents and Trademarks

US005259033A

United States Patent [19]

Goodings et al.

[11] Patent Number: 5,259,033

[45] Date of Patent: Nov. 2, 1993

[54] HEARING AID HAVING COMPENSATION FOR ACOUSTIC FEEDBACK

[75] Inventors: Rupert L. A. Goodings; Gideon A. Senensieb, both of Cambridge; Philip H. Wilson, Surrey, all of United Kingdom; Roy S. Hansen, Dragor, Denmark

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[58] Field of Search 381/68, 68.2, 68.4, 381/83, 84, 106, 71, 93

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"Measurement and adaptive suppression of acoustic feedback in hearing aids", by D. K. Bustamante et al, ICASSP, Glasgow, Scotland, May 23-26, 1989, pp. 2017-2020.

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Assistant Examiner—Jason Chan

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[57] ABSTRACT

A hearing aid includes a filter in an electrical feedback path, the characteristics of which filter are calculated to model acoustic coupling between the receiver and microphone of the aid. A limiter is inserted in the main electrical pathway between the microphone and the receiver to provide stability in the presence of sudden sound bursts. A noise signal is injected continuously into the electrical circuit and is used to adapt the characteristics of the filter to accommodate changes in the acoustic coupling. The level of the noise signal can be varied to match changes in residual signal level to maintain signal to noise ratio and to optimize rate of adaption commensurate with satisfactory hearing function while the noise itself is unobtrusive to the user.

17 Claims, 6 Drawing Sheets

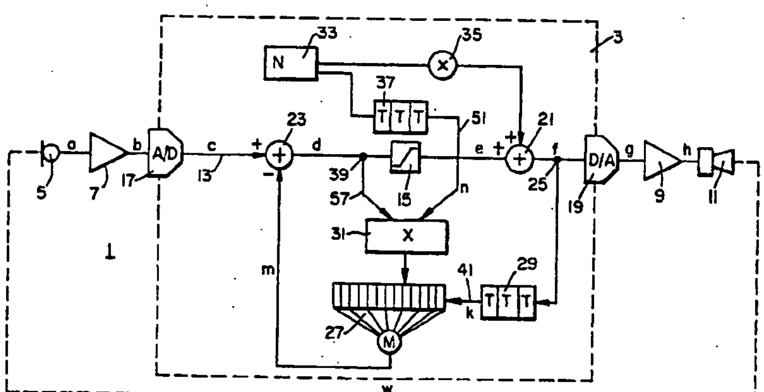

Fig.1.

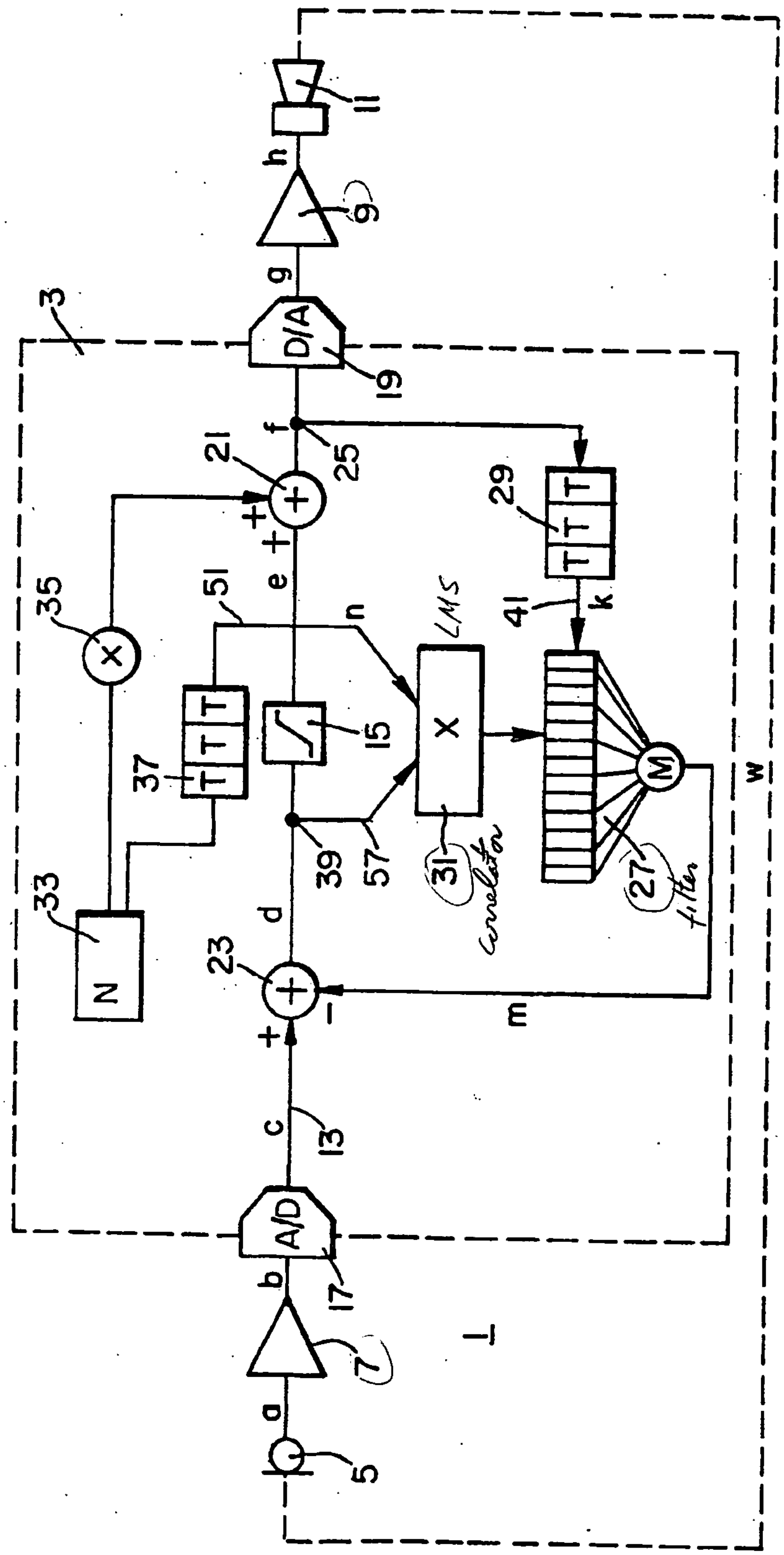
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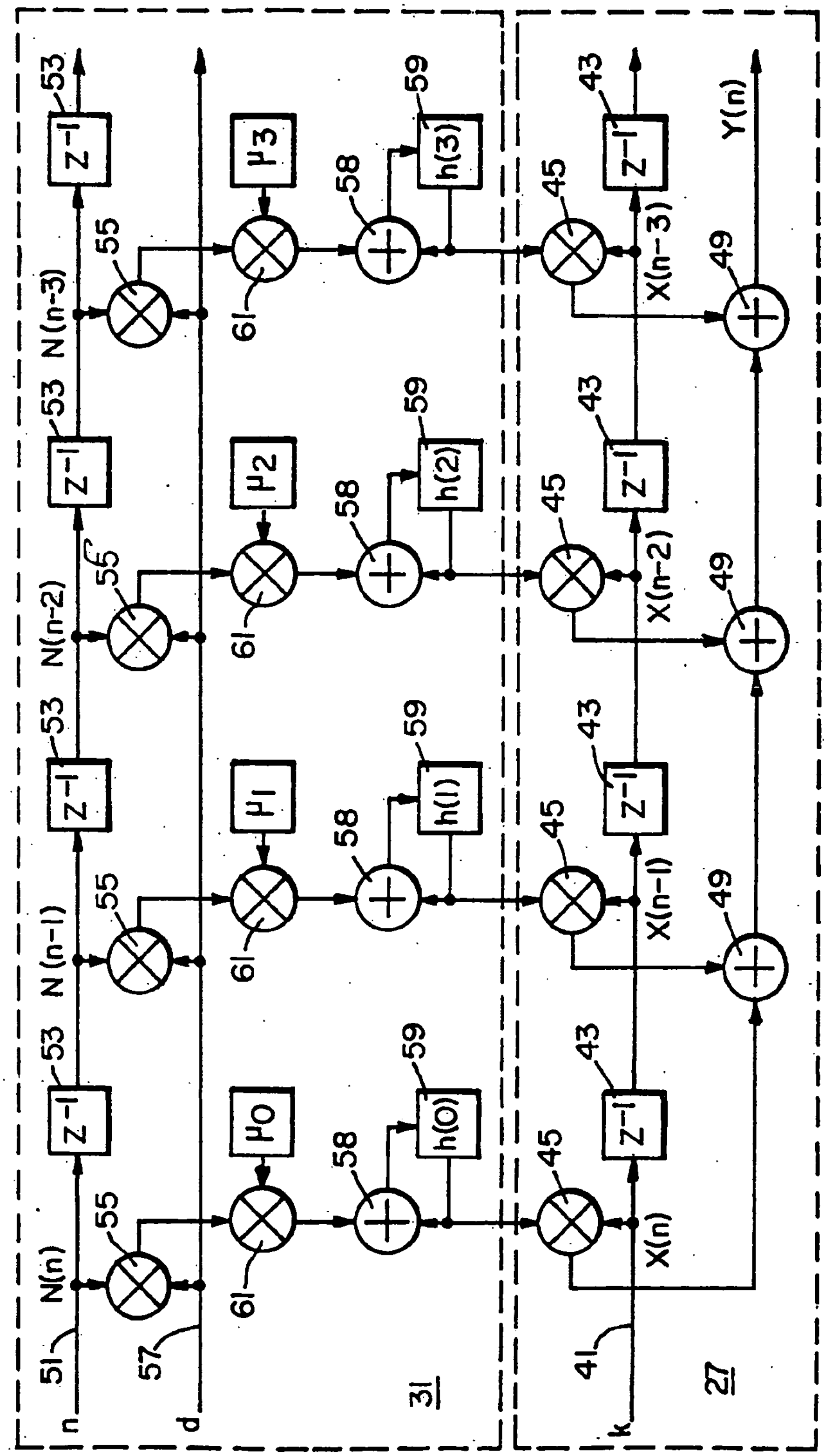
Fig. 2.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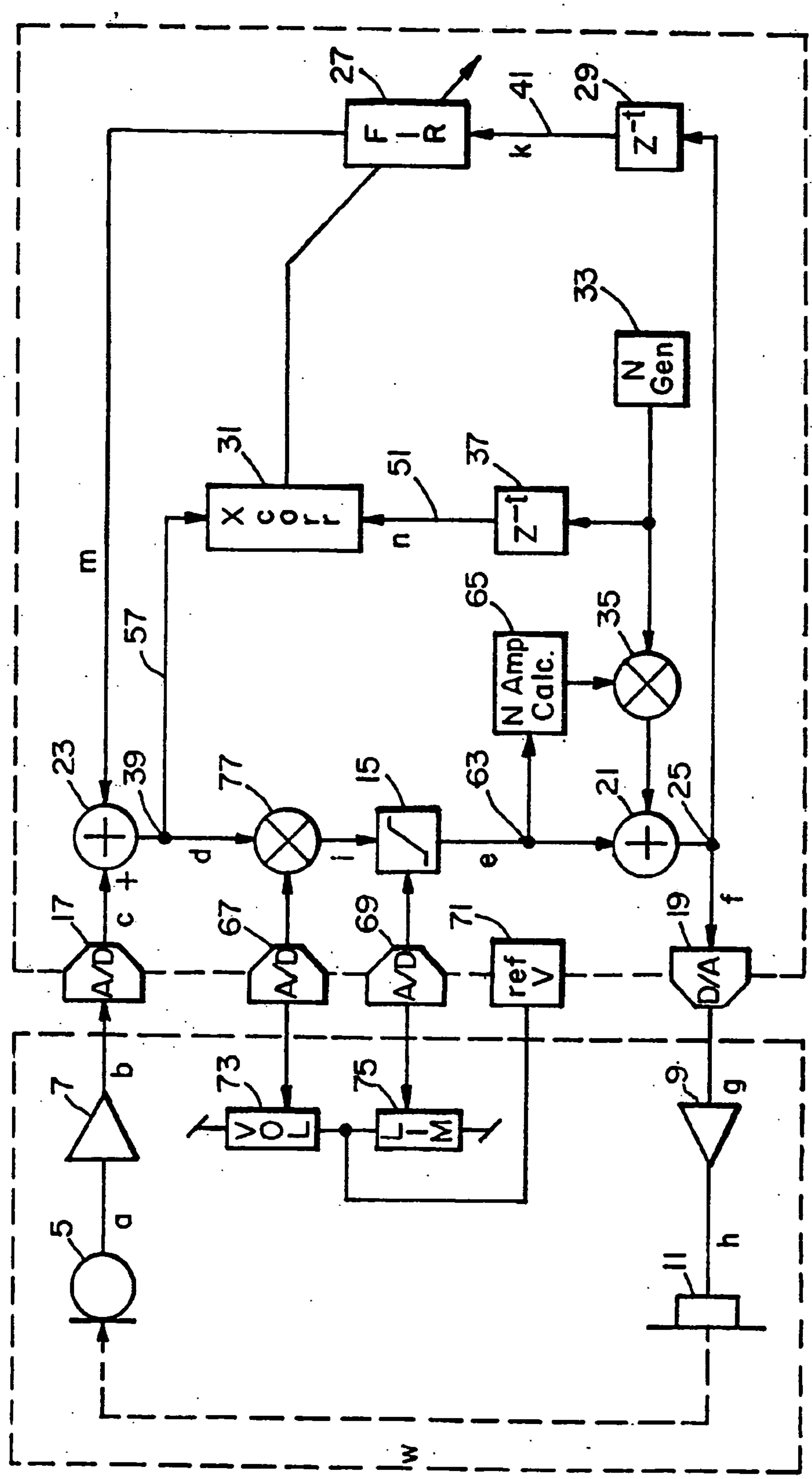
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Fig.3.

Fig.4.

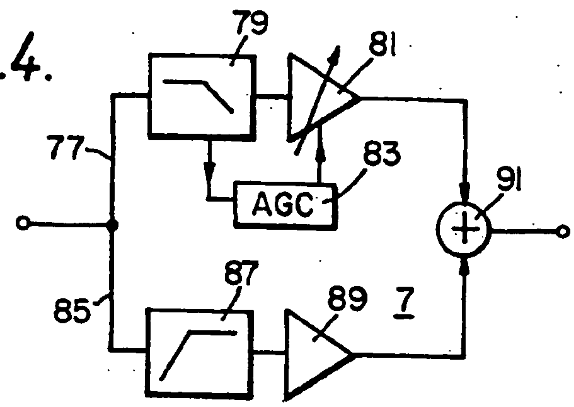


Fig.5.

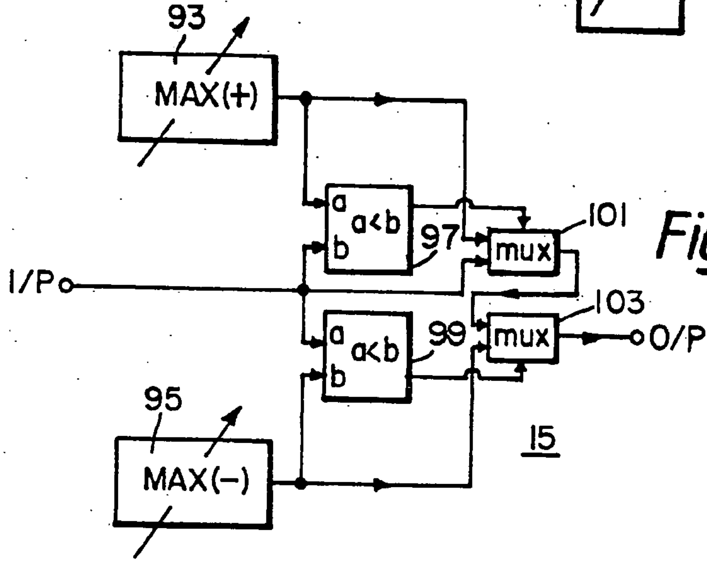


Fig.6.

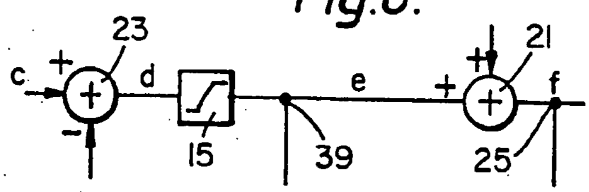


Fig.7.

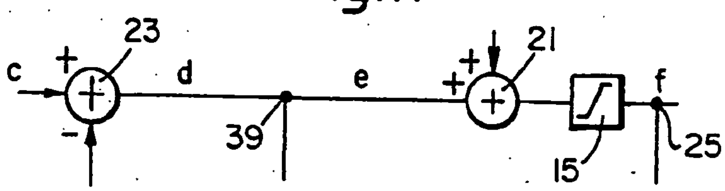


Fig.8.

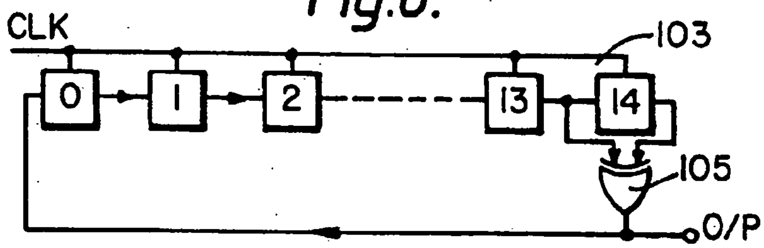


Fig.9.

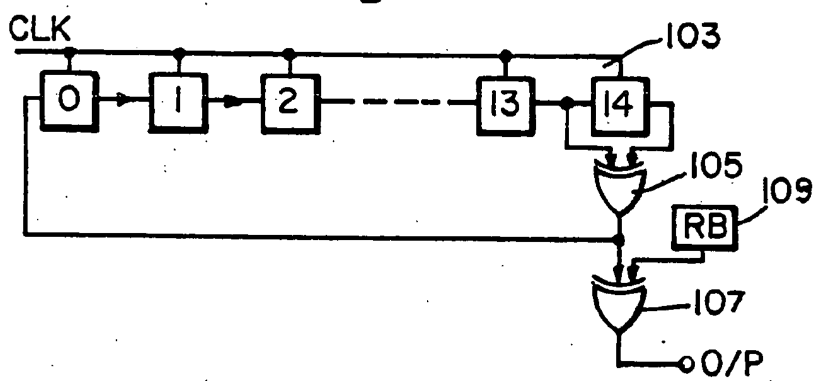


Fig.10.

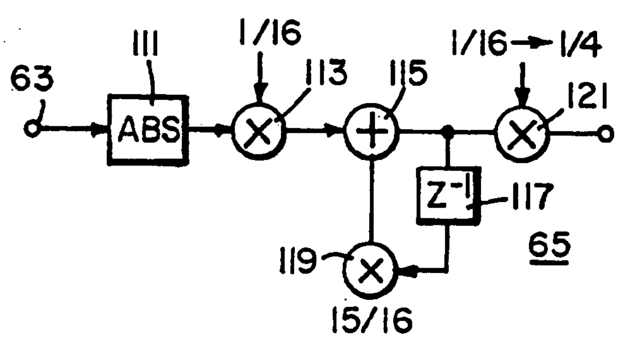
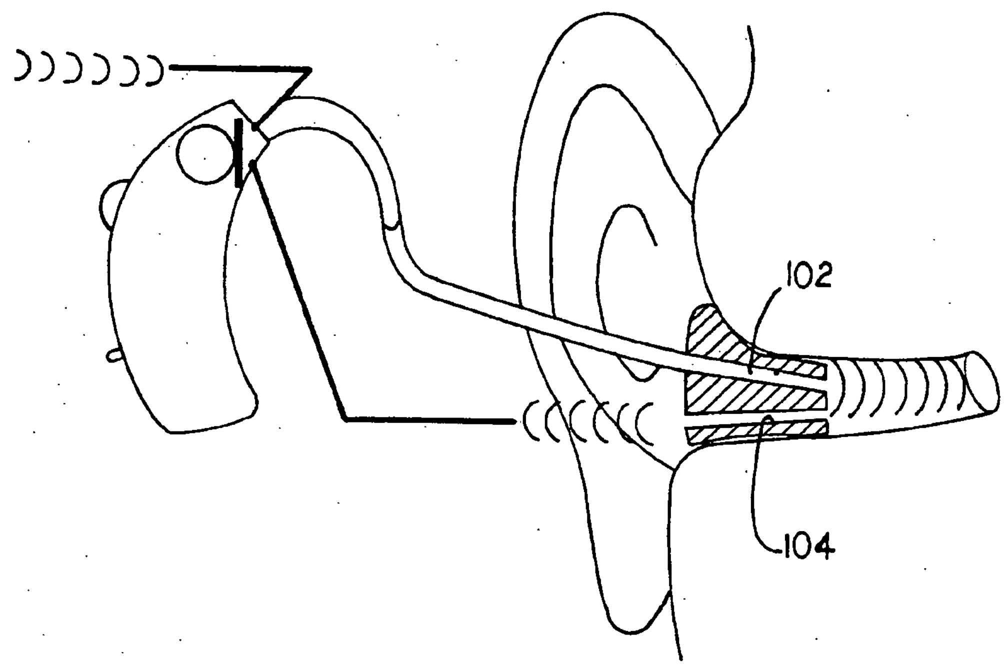


Fig.11



HEARING AID HAVING COMPENSATION FOR ACOUSTIC FEEDBACK

This application is a continuation of application Ser. No. 07/567,370, filed Aug. 14, 1990, now abandoned.

BACKGROUND OF THE INVENTION

1. Technical Field

The present invention concerns improvements in or relating to hearing aids and more particularly hearing aids such as for example "in-the-canal" (ITC), "in-the-ear" (ITE), and "behind the ear" (BTE) hearing aids which share the common feature that the microphone and receiver components thereof are mounted on a single body and are closely spaced.

A number of mechanisms can contribute to the acoustic feedback that occurs in these aids. In all cases there will occur an unavoidable acoustic or mechanical coupling through the body or the inner volume of the hearing aid. Acoustic feedback will also occur as sound is directed by reflection and is channelled from the receiver to the microphone. Thus in the case of "behind-the-ear" aids sound can be reflected by the surrounding head structures and in the case of "in-the-canal" and "in-the-ear aids" acoustic coupling can occur through an unintended aperture around the periphery of the mould of the hearing aid.

In the case of "in-the-ear" hearing aids, venting is included as a means for reducing occlusion—that is to say, to avoid a blocked ear sensation. A further benefit of including venting is that it allows the direct input of unamplified low frequency sound into the ear—an important benefit for those patients that suffer a high frequency loss with only minor loss at low frequency and who have therefore no need for amplification in this lower frequency range. There is also available a wide range of vent sizes. Whilst larger diameter vents are more effective for reducing occlusion than vents of smaller diameter, they also give rise to an increase in acoustic feedback. Vent sizes are thus limited by the attendant need to avoid excessive acoustic feedback. Large sizes of vent now are used infrequently and are limited in application to aids using only lower gain amplification.

A number of problems are associated with acoustic feedback. The most commonly considered is the occurrence of oscillation. This occurs at high gain level and is usually persistent. At gain settings just below those that cause such oscillation "ringing effects" are perceivable. This latter causes further unpleasant sensation and reduces the intelligability of speech and other wanted sound. An additional effect is that the acoustic feedback contributes to gain dependant perturbations in the frequency response of the hearing aid. Such uncontrolled frequency response effects occur at gains well below the onset of oscillation and can cause quite discernable, and sometimes considerable, distortion in the perceived sound, and accordingly do reduce the effectiveness of the hearing aid in meeting prescribed characteristics.

2. Reference to Prior Art

It is difficult to avoid the occurrence of acoustic feedback. Thus in the case of "in-the-ear" aids leakage can be reduced by making the ear mould a tighter fit. This however is hard to achieve and is far from ideally comfortable for the user. It is also not possible to eliminate internal acoustic feedback. In both cases however it is possible to reduce the level of acoustic feedback by careful positioning of the components, particularly the direction and position of the microphone port.

Since the elimination of acoustic feedback is not possible, effort has been directed to suppress the occurrence of oscillation. The frequency response of hearing aids means that oscillation usually occurs at frequencies in the range 1 to 4 kHz and the exact frequency is set by the normal Nyquist conditions (i.e. loop gain that is greater than zero decibels and zero phase difference between wanted signal and echo).

Oscillatory conditions may be suppressed by the user turning down the volume control, i.e. reducing gain, once oscillation has occurred. This however is generally inconvenient to the user. Therefore techniques have been developed for preventing the occurrence of oscillation. Thus the electronic gain of an aid has been reduced for frequencies where oscillation is likely to occur. This however means that the prescribed high frequency response for a deaf user would be compromised deliberately at these high frequencies which are usually in the range 1 to 4 kHz and also quite near to the natural resonant frequency of the ear canal of about 3 kHz. Also receivers have been designed to reduce their electro-acoustic gain particularly at the resonant frequency of the receiver where a relative gain figure of 10 decibels or more can occur. However, this peak is normally desirable so as to reproduce the natural canal resonances of a normal ear.

In all cases of oscillation avoidance considered above it will be noted that a design compromise is necessary and that in consequence such hearing aids can only offer a sub-optimum performance to the user.

Other corrective techniques have involved the detection of the onset of oscillation and the provision of cancellation or other compensation in response. Thus a detected signal has been used as a means of controlling an automatic gain stage. An alternative technique is to use the detected signal to control the generation of an internal oscillation and to use the latter to cancel the unwanted oscillation. This technique has to be adaptive since the oscillation frequency and amplitude will vary according to changes in external conditions. Furthermore the detected signal has been used to adjust the centre frequency of a notch filter, that is to say a filter with a frequency selective reduction in gain centered at or near the oscillation frequency.

In the above techniques involving oscillation detection however no compensation is provided for the effects of acoustic feedback at those frequencies remote from the oscillation frequency.

An alternative approach to the above is proposed in U.S. Pat. No. 4,783,818. As proposed the effects of acoustic feedback are compensated by electrical negative feedback. The negative feedback path includes a filter the characteristics of which are modeled on those identified inter alia for the acoustic feedback path itself. The communication system described, which may be for example a hearing aid, is constructed to have two time consecutive modes of operation: an ordinary operational mode, and an identification mode. In the ordinary operational mode the compensated electrical signal is fed through an amplifier and thereon used to energise the receiver. In the identification mode, which may be selected e.g. at turn-on or in response to an automatically sensed threshhold change in amplifier gain, the amplifier is decoupled from the circuit and a correction circuit is substituted. The correction circuit includes a source of noise for example pseudo-random-

binary-sequence signals and parameters associated only with acoustic feedback are identified and used to define the transfer function of the filter that is then subsequently used during ordinary operation. It is a disadvantage of this construction however that during the identification mode of operation the hearing aid behaves as a generator of acoustic noise and does not provide any effective hearing aid operation at all. Also ordinary operation is only satisfactory so long as the acoustic feedback remains substantially the same as that for which the parameters are identified. The proposed construction therefore is not entirely satisfactory for hearing aid application.

This type of approach has also been considered experimentally. See the following thesis entitled "Digital Suppression of Acoustic Feedback in Hearing Aids" by Leland C. Best, University of Wyoming, May 1985. In this are described several experiments on an adaptive digital filter applied as a continuously adaptive feedback path to compensate for acoustic feedback in the hearing aid. A digital random number generator using the congruence method is employed as the source of noise which is injected into the acoustic path for deriving adaptively the filter parameters.

The experimental apparatus described in this thesis does not form a practical hearing aid. This is principally because of the problem of non-linearities in the feedback, as described in Chapters IV and VI of this reference. Such non-linearities become manifest when the output signal momentarily exceeds the linear region of the response of the output transducer. Under these circumstances the digital filter is no longer able to produce an accurate replica cancelling the feedback signals. Since the digital filter would generally be employed to enhance the gain of the hearing and, it would be quite common for both the acoustic path and the digital filter path each to be unstable when separate, only stabilising each other by producing accurate replicas of each others feedback signals. At the onset of non-linearity as described the feedback signals in the two parts no longer cancel each other and both parts become unstable. The hearing aid therefore becomes unstable, and may remain so indefinitely.

In Chapter IV of this reference, this problem is solved for experimental purposes by detecting that this situation is likely to have occurred, and silencing the transmission of output signal until this system recovers. This is performed for example by ramping down the amplifier gain to zero. See pages 26 to 28 of this reference. Such a solution would not be acceptable in practice however since it would result in fairly frequent interruptions of the amplified sound.

SUMMARY OF THE INVENTION

The present invention is intended to provide a remedy to the problems aforesaid.

It is an object of the present invention to provide compensation for acoustic feedback occurring due to unavoidable acoustic coupling between the receiver and microphone of the hearing aid.

It is a further object to provide operational stability for the hearing aid in the presence of abrupt and transient increases in the level of ambient sound.

It is a preferred object of the invention to provide adaptive change in such compensation to accommodate temporal changes arising in the acoustic coupling between the receiver and the microphone.

It is a further preferred object of the invention to provide such adaptive change without any interruption of hearing aid function.

It is yet a further preferred object of the invention to provide such adaptive change in continuous response to the temporal changes arising in the acoustic coupling aforementioned.

According to the present invention, there is provided a hearing aid in which acoustic feedback between receiver and microphone is electrically compensated by applying an electrical feedback signal to an electrical path including an amplitude limiting device between the microphone and the receiver, the amplitude limiting device being arranged such that the receiver shall not enter its non linear region of operation which is outside of the amplifier-receiver upper and lower linearity limits, the electrical feedback signal being derived by extracting an electrical response signal from the electrical path after the output of the amplitude limiting device and passing the same through a filter the characteristics of which are calculated to model acoustic coupling between the receiver and the microphone.

The present invention provides for practical application of a feedback cancellation by the deliberate inclusion in the electrical output path of a device which has such performance characteristics as will ensure that the signal is prevented from reaching the non-linear region of the transducer response. The device is included in the circuit in such a location as it can effect the signal applied both to the output transducer and to the cancellation network so as to ensure that the signal input to the cancellation network is at all times representative of the acoustic signal emanating from the transducer. Both the acoustic path and cancellation path thus operate in their respective linear regions and non linear oscillations are avoided. The need to interrupt the transmission of amplified sound is thereby eliminated.

BRIEF DESCRIPTION OF THE DRAWINGS

In the drawings accompanying this specification:

FIG. 1 is a block circuit diagram of a hearing aid having provision for adaptive electrical feedback;

FIG. 2 is a block circuit diagram of adaptive components used in the above modified hearing aid;

FIG. 3 is a block circuit diagram of a hearing aid a modification of the aid shown in FIG. 1;

FIG. 4, is a block circuit diagram for an amplifier including automatic gain control;

FIG. 5 is a block circuit diagram for a limiter such as that used in the circuits of FIGS. 1 and 3 above;

FIGS. 6 and 7 are block circuit diagrams illustrating alternative positions for the limiter used in the aid of FIGS. 1 and 3 above;

FIGS. 8 and 9 are block circuit diagrams of a pseudo-random-binary-sequence generator and a variant thereof;

FIG. 10 is a block circuit diagram of a noise level control circuit for use in the hearing aid shown in FIG. 3; and

FIG. 11 is a schematic view of a vented hearing aid for use with the present invention.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

Embodiments of the invention will now be described and reference will be made to the drawings. The description that follows is given by way of example only.

In FIG. 1 there is shown a hearing aid 1 which but for the inclusion of an adaptive circuit 3 is otherwise of conventional design and comprises an acousto-electric transducer—i.e. a miniature microphone 5—a preamplifier 7; a main amplifier 9; and an electro-acoustic transducer—i.e. a receiver 11. The preamplifier 7 may be constructed as shown in FIG. 4 to provide a compensation for excessively loud speech signal such as that eminating from the user of the hearing aid. In one path 77 of this amplifier there is included a low pass filter 79 and a first amplifier 81. The gain of this first amplifier 81 is controlled by means of an automatic gain control circuit 83. In the other parallel path 85 of the preamplifier 7 there is included a high pass filter 87 and a second amplifier 89. The signals from each of the amplifiers 81 and 89 are summed at a summing node 91. In the connective electrical path 13 between the preamplifier 7 and the main amplifier 9, a limiter 15 is inserted so as to prevent the electrical signal reaching an amplitude level exceeding the amplifier-receiver upper and lower linearity limits of each of the main amplifier 9 and the receiver 11. In the arrangement of conventional components and limiter thus far described the acousto electric gain of the microphone 5, the gain of the preamplifier 7, the gain of the main amplifier 9, and the electro-acoustic gain of the receiver 11 are all designed to have a suitable spectral response so that the insertion gain of the hearing aid conforms (matches) as near as is possible to prescriptive requirements.

For general convenience the adaptive circuit 3 has been constructed using an all digital implementation. The construction of the adaptive circuit 3 however is not restricted to the details of the design shown and mixed digital-analogue and/or analogue implementations are not precluded.

The input of the adaptive digital circuit arrangement 3 is interfaced to the preamplifier 7 by an analogue to digital converter 17. At its output the digital adaptive circuit 3 is interfaced to the main amplifier 9 by a digital-to-analogue converter 19. The limiter 15 is also of digital construction (FIG. 5) and is located in the connective path between the analogue-to-digital converter 17 and the digital-to-analogue converter 19. The signal at the input to the limiter 15 is compared in parallel by means of comparators 97 and 99 and the output signal of each of these comparators 97 and 99 is used to operate serially connected multiplexers 101 and 103. When the signal is within the set limits the signal is passed through the two multiplexers 101, 103 to the output. If however one or other of the limits is exceeded the signal is locked and one or other of the limit value signals passed to output by the multiplexers 101, 103. A summing node—a digital adder 21—is interposed in the electrical connective path between the limiter 15 and the digital-to-analogue converter 19. This digital adder 21 is to provide a node for the injection of a noise signal N as will be described hereinbelow. Of course, it will be recognized that the limiter upper and lower linearity limits of limiter 15 must lie within the amplifier-receiver upper and lower linearity limits, and the limiter upper and lower linearity limits differ from the amplifier-receiver upper and lower linearity limits by an amount at least equal to the level of the noise signal N. A subtractive node—a second digital adder 23—is inserted in the main path between the analogue-to-digital converter 17 and the limiter 15. This latter digital adder 23 is provided as a means for introducing electrical feedback which also will be described hereinbelow.

The normal signal path for wanted signals from the microphone 5 to the receiver 11 is the direct path a-b-c-d-e-f-g-h as shown. It will be noted that the path elements a, b, g and h are analogue and thus provided by single conductors. Direct path elements c, d, e and f however are all digital and will thus comprise each a number of parallel conductors. Each path element could have for example between 8 and 12 parallel conductors according to the bit resolution of the analogue-to-digital converter 17.

Electrical feedback is extracted by means of a tap 25 on section f of the main path, that is to say the section between the digital summer 21 and the digital-to-analogue converter 19, i.e. the electrical feedback signal taken from this tap includes an injected noise signal component. The tapped electrical feedback signal is passed through an adaptive filter 27, as shown here a finite impulse response (FIR) filter, and after passage through this filter 27 it is applied to the subtractive digital adder 23. For convenience, in this arrangement the digital signal from the tap 25 is passed through a delay line 29 before reaching the filter 27. The delay time of this line 29 is designed to model the minimum acoustic transit time between the receiver 11 and the microphone 5.

The inclusion of this delay line 29 is not essential but it avoids redundancy in filter and correlator stages. The finite impulse response filter 27 is adaptively controlled by multiplicative coefficients derived using a correlator 31. The correlator 31 looks for a correlation between the injected noise signal N and any noise component in the residual signal formed at the output of the subtractive node, i.e. the digital adder 23. The injected noise signal N is generated using a source 33 and is injected into the main signal path using the summing node, digital adder 21, after attenuation via a multiplier 35. The noise signal is also connected to a reference input of the correlator 31 and is passed to this via a second delay line 37 which also inserts a time delay corresponding to the minimum acoustic transit time already described. The residual signal which is applied to the signal input of the correlator 31 is taken from a tap 39 on the digital section d of the main path, that is the main path between the subtractive node digital adder 23 and the digital limiter 15.

A copy of the electrical output signal at tap f is thus passed through the delay 29 into the adaptive Finite Impulse Response (FIR) filter 27. This filter is arranged to produce the adaptive feedback signal.

This adaptive feedback signal is subtractively summed with the input signal. In ideal operation the adaptive feedback signal will be arranged to be equal in all respects to the unwanted acoustic feedback that occurs as a result of the mechanisms described earlier (the acoustic feedback is denoted by a dotted line w in FIG. 1).

If the adaptive feedback signal and the input signal to the digital adder 23 are equal then the output of the digital adder 23 contains no residual feedback because the two components will exactly cancel each other.

In order to adapt the filter 27 so that it produces an exact replica of the electrical signal corresponding to the acoustic feedback, the noise signal N, after attenuation via the multiplier 35, is added to the output signal and is used to excite the receiver 11. Note that the noise signal appears in both the internal (adaptive) and external (acoustic) feedback paths, so that in ideal operation noise feedback will be completely cancelled at the digi-

tal adder 23 so that the residual signal on line d contains no injected noise.

It will be noted that the noise signal N injected into the main signal path at the summing node 21 passes through the digital-to-analogue converter 19 and thence through the main amplifier 9 and the receiver 11. At the receiver 11 it is converted to produce an acoustic signal. This signal is superimposed upon the wanted signal. The level of the noise signal produced at the output of the multiplier 35 is therefore so chosen that the acoustic level of the output of the receiver 11 is unobtrusive to the user. Since correlative techniques are used and these are very sensitive, it is possible to use an extremely low level of noise signal, indeed one that is little higher than (i.e. within a few dB above) the white noise level of the hearing aid itself. For practical use a very efficient adaptation behaviour can be obtained by using a constant signal/noise ratio. The level of injected noise is continuously adjusted to be a certain number of dB lower than the signal and therefore unobtrusive to the ear. A high level of noise relative to the signal gives a very fast adaptation. This instantaneous signal/noise ratio can be say between 12 and 24 dB with 12 dB being suitable for persons with severe hearing losses requiring very high gain. 24 dB would result in somewhat slower adaptation, but with a noise level practically imperceptible to a person of near normal hearing requiring more moderate gain levels.

In practice, the internal (adaptive) feedback and the external (acoustic) feedback do not completely cancel and a small amount of noise (and other feedback components) appear as finite residual signal. The noise component of this residual signal is detected by the correlator 31 which usually operates continuously, looking for long term agreement between the residual signal and a delayed version of the generated noise. The output of the correlator is an estimate of the residual noise in the signal, and it is used to directly adapt the filter 27. The adaption sign is choosen to adapt the filter 27 to produce lower residual noise so that the system tries to converge to a situation where the noise is cancelled. The dynamic nature of the environment, and practical algorithm limitations mean that perfect convergence is not generally achieved. However, given careful optimisation high levels of cancellation are possible.

By arranging the noise to provide a satisfactory spectral characteristic, that is a flat level over a bandwidth comparable to that of the hearing aid, the adaption of a system for minimum noise components in the residual noise signal also corresponds to minimum residual feedback of all other signals. It has here been shown that pseudo-random-binary-sequence p.r.b.s. signals of appropriate bit repetition lengths can serve as satisfactory noise signal. Such pseudo-random-binary-sequence signals can be readily generated (FIG. 8) using a clocked shift register 103 with multiple feedback via an exclusive OR-gate 105. Such configured registers produce signals with a pattern repeating after every 2^M-1 bits where M denotes the number of register stages. Satisfactory performance has been obtained for sequences of length 127 bits and 37,267 bits (i.e. using registers having 7 and 15 stages respectively). Other spectrally balanced sequences (including other lengths of p.r.b.s.) are also expected to work.

Whatever the choice of noise signal it should have a low autocorrelation over any time span which is of the same order of magnitude as the adaption time constant, typically a second. The pseudo-random bit sequence generator described has the advantage of being simple to implement and also is suitable for miniaturisation. It is noted however that it is a disadvantage of such a simple device, particularly when a low order pseudo-random bit sequence is employed, that it has a low repetition period. This can be a problem when the ambient acoustic signal is highly periodic—for example a long duration constant frequency sine wave signal. Spurious correlations with the injected signal may then occur and cause the adaption process to wander, leading to undesirable audible effects. These effects can be eliminated by providing a further randomisation of the pseudo-random bit sequence. This is shown for example in FIG. 9 where the signal at the output of the p.r.b.s. generator 103, 105 is gated by a second exclusive OR-gate 107 whose other input is connected to a source 109 of a randomising binary signal, for example it may be connected to the least significant digital output of the analogue digital converter 17 of the hearing aid. This has the effect of randomising the bit sequence further, and thus of eliminating the wandering effect.

It will be noted that by reducing cancellation of feedback over a broad range of frequencies, not just a small range of frequencies where oscillation can occur, all of the effects of the acoustic feedback can be removed. Thus it would be normally ensured that the noise signal has a flat level characteristic over the range 300 Hz to 7 kHz, the conventional bandwidth of a hearing aid.

Where only avoidance of oscillation is requisite however a noise signal having a flat spectral characteristic over a more limited range covering the expected range of oscillation frequencies normally would be adequate.

The correlator 31 and FIR filter 27 are shown in more detail in FIG. 2

The FIR filter 27 is of standard digital filter architecture. It comprises thus: a delay line 41 having a number of delay stages 43; a first multiplier 45 preceding the first one of the delay stages 43 and a multiplier 45 following each delay stage. The multipliers 45 are all connected in parallel to the input of a digital adder 49.

The input signal on digital lines k of the output of the delay line 29 is passed through the series of delay stages 43 to provide a series of sequential samples x (n), x (n−1), x (n−2) . . . etc where x (n) is the most recent digital sample of the signal. Each sample is delayed one period of the master clock operating the analogue-to-digital converter 17 and the digital-to-analogue converter 19. Typically, for an in-the-ear aid (ITE) the upper band limit would be designed to be on the order of 7 kHz. For this the master clock frequency would need to be at least 14 kHz (Nyquist) and in practice a frequency of at least 20 kHz would be chosen. For a behind-the-ear aid the band width would be in any case somewhat lower since the communicative tubing used in its construction acts as a low pass filter. In such case a lower master clock frequency of say 10 kHz would then be more appropriate. A master clock oscillator including a switched capacitor filter could be used and could be preset to provide a master clock frequency of either 10 kHz or 20 kHz as requisite. The common circuit design may then be implemented. It is noted that although the higher clock frequency of 20 kHz could also be used for the BTE aid, by using a lower sample rate advantage is taken of the fact that the correlation is performed over a greater time span for a given number of delay stages. Thus performance can be improved or alternatively a similar performance achieved using a smaller number of correlator stages. The signal samples

are individually multiplied by a series of multiplicative coefficients h(0), h(1), h(2) . . . , etc., and the resulting products are summed by the digital adder 49 to produce a single output y (n). This operation is summarized by the formula:

$$y(n) = \sum_{m=0}^{N-1} \{h(m) \cdot x(n-m)\}$$

In this arrangement each of the coefficients h(m) is updated on each cycle of the master clock and a new output y (n) is calculated.

Adaption operates by adjusting the values of the coefficients h(m) in a controlled manner. A correlator 31 designed to do this is also shown in FIG. 2. This correlator 31 is designed to adapt the filter 27 according to the Widrow-Hoff algorithm (see B. Widrow et al., "Stationary and non-stationary learning characteristics of the LMS adaptive filter," Proc. IEEE Volume 24 pages 1151-1162, August 1976). In applying this algorithm each coefficient h(m) is adjusted every cycle. The adjustment is a simple increase or decrease in the value of the coefficient h(m) the magnitude and the sign of which are determined by the action of the correlator. Each coefficient h(m) is accumulated independently.

The correlator 31 comprises a delay line 51 having a number of single bit delay stages 53. The number of these stages is the same as the number of stages 43 incorporated in the FIR filter 27 just described. The input to the delay line 51 and the output from each of the single bit delay stages 53 is connected to the reference input of a corresponding digital multiplier 55. The second input to each of the digital multipliers 55 is connected to a common set of digital lines 57. The delay line 51 is connected to receive noise signal N from the noise source 33 and delay line 37, whilst the common set of lines 57 is connected to tap 39 so as to receive the residual signal. The output of each multiplier 55 passes to an adaptation rate scale factor multiplier 61 and the product term is either added, by an adder 58, to an accumulated coefficient held in an accumulator . register 59. The amplitude of this added term differs from the amplitude of the residual signal by an adaption-rate scale factor. The sign of this term, i.e. whether it is added or subtracted, depends upon the signs of the noise and residual signal being the same or different respectively.

The process is simplified by combining the long term averaging that is required for correlation into the coefficient accumulators. This is realised by extending the resolution of the coefficient accumulators 58-59, so that several successive correlation inputs are required before the accumulated input becomes significant to the filter 27. Thus in the construction shown each coefficient accumulator 59 is 24 bits in length of which only the most significant 12 to 14 bits are used to define the multiplicative coefficients h(m) used in the FIR filter 27. The ratio of maximum correlator input to filter significance determines the adaption rate. It is a property of the chosen adaption algorithm described above that the adaption will vary as a function of signal level. In this preferred arrangement the adaption is continuous since the multiplicative coefficients are updated upon each master clock cycle. The adaption rate should therefore be carefully chosen to get an acceptable compromise between the rate of adaption from cold (or following an abrupt change in sound signal or in ambient conditions) and the degree of convergence (i.e. stability of the system) when after a settling period the conditions are stable. It is a drawback of the correlator thus far described that when it is designed to optimise the degree of convergence, i.e. long term stability, the settling period from start up can be somewhat excessive. As an improvement therefore a modification is provided to impose additional variation of the adaption rate. Thus in FIG. 2 the correlator 31 is modified by the insertion of a second multiplier in each stage between the first multiplier 55 and the corresponding coefficient accumulator 59. Each second multiplier 61 has an input to receive a scaling coefficient μ. The signed residual signal produced at the output of each first multiplier 55 is thus scaled by a multiplicative factor μ before it is passed on to the corresponding coefficient accumulator 59 for each consecutive stage of the correlator 31. Various techniques are thus possible to optimise the above trade-off between rate of adaption from cold (or an abrupt change in conditions) and the degree of convergence when conditions are stable. The preferred implementation thus uses an imposed variable rate of adaption. In practice it has thus been found that better performance is achieved by imposing a high rate of adaption when the hearing aid and thus correlator 31 are first switched on (i.e. from cold) followed by a lower imposed rate of adaption after a predetermined settling period, the latter for achieving long term stability. To implement this the multiplicative scaling coefficients μ are first assigned a common high value and after a predetermined settling time the coefficients μ are abruptly changed to a common second lower scaling value. Circuit components for generating the scaling coefficients μ thus may include a time measuring detector e.g. a counter that is triggered by the master clock. This then will accommodate the need for rapid adjustment of the multiplicative coefficients μ that are fed to the FIR filter 27 immediately following start-up. In order to accommodate not only rapid adjustment of the coefficients h(m) following start up but also rapid adjustment of the coefficients h(m) following an abrupt change in conditions long after start up, the counter could instead be triggered using a discriminator that is responsive to the level of the residual signal. Such implementations have proved to be altogether quite adequate. In the foregoing modification to the correlator 31 the scaling coefficients μ are changed step-wise between two values, a high value and a low value respectively. Other techniques of optimisation however are not precluded. In a more complex variant using an alternative design of counter the scaling coefficients μ may be changed from a high value to a low value by gradual transition through a series of decreasing values. In both the foregoing modifications discussed the scaling coefficients μ for each of the stages of the correlator 31 are changed in unison. It is also noted that when conditions are very stable no adaption would be required and updating of the accumulated coefficients could be inhibited by setting the scaling coefficients μ to zero for all stages of the correlator 1.

Instead of the foregoing, the adaption rate could be varied by changing the level of the injected noise. This may be implimented, for example, by controlling the level of the injected noise to vary as a function of the level of the audio signal. Provided that this is done keeping the ratio of signal to noise more or less constant, the net noise output whilst audible would be quite unobtrusive. Further details will be given below.

The embodiment and its variants described above may be implemented as part of the internal circuitry of a hearing aid. The Widrow-Hoff algorithm is highly efficient. The circuitry therefore is particularly appropriate to the construction of "behind-the-ear" "in-the-ear" and "in-the-canal" aids where low power operation is required. The present invention may be used in a vented hearing aid, as shown in FIG. 11, having first and second passageways 102 and 104, the second passageway 104 being open at both its ends to provide venting for the ear when the molding of the hearing aid is fit in the ear.

The limiter 15 described above is located as shown in FIG. 1 between the correlator input tap 39 and the injection node 21. This position is optimum. However as an alternative to the above it could be located at other positions between the feedback injection summing node 23 and the feedback input tap 25. These alternative two positions are shown in FIGS. 6 and 7. In FIG. 6 it is shown located between the feedback signal injection summing node 23 and the correlator input tap 39. For this location however signal clipping results in a loss of information about the noise that has been added and this results in a slow down of the adaption. Similarly when the limiter is located between the noise injection summing node 21 and the feedback input tap 25 the injected noise is limited and again the adaption is slowed down.

Further refinements of this hearing aid are shown in FIG. 3. This refined embodiment includes a user controlled volume control and also provides for external variation of the limitation levels for the limiter. Provision is also made for signal to noise level control and also for preset adjustment by the user.

In a practical hearing aid it is desirable to have a volume control which can be user controlled. This can be placed in the microphone amplifier or before the output amplifier, but in both cases the adaptive filter must change its coefficients when the volume control is adjusted. If however the volume control is placed after the subtraction of the anti-feedback signal and before the limiter neither the maximum output nor the coefficients need to change when the volume control is adjusted. In the adapted arrangement shown in FIG. 3 a multiplier 77 is inserted between the correlator input tap 39 and the limiter 15. The control for the multiplier 77 is provided by a user controlled volume control potentiometer 73 and an A/D converter 67.

The limits for the limiter 15 could also be user controlled. This is desirable because the limiter determines the maximum sound pressure level which is able to occur in the user's ear. This is a significant feature because the maximum output can be reduced or altered without reducing or altering the gain of the amplifiers. As shown in FIG. 5 the maximum positive and negative limits 93, 95 for the limiter 15 are provided by a user controlled potentiometer 75 and an A/D converter 69. In the simple arrangement shown the volume control and limit control potentiometers 73, 75 are connected to a common source 71 of a reference voltage.

One or more filters would normally be included to modify the spectral response according to prescriptive requirements. Preferably in the hearing aid circuit shown in either FIGS. 1 or 3 this is implemented digitally using one or more FIR or IIR filters. It is preferable to locate such a filter or filters between the limiter 15 and the feedback cancellation signal node 23. Such filters would usually introduce added delay, but if placed in this preferred location it is not necessary to add extra delays in the feedback cancellation loop. This would be necessary if the filter were placed to immediately precede the digital to analogue converter 19.

Due to the limited length of the feedback cancellation FIR filter used in the embodiments described, the possible cancellation grows with the frequency, and is therefore most effective in hearing aids with a relative high gain at the higher frequencies. With some kinds of hearing loss it is desirable to reduce the high frequency gain, because the user's residual hearing at loss of frequencies is useless. Inevitably this kind of filtering will reduce the effect of feedback cancellation. A solution is to reduce the sampling frequency. This requires however that the anti-aliasing filter cut-off frequency tracks the sampling frequency. This could be done with a switched capacitor filter coupled to the A/D converters 17, 67 and 69.

As already mentioned above the noise level may be controlled to provide optimal rates of adaption. As shown in FIG. 3 the multiplier 35 following the noise generator 33 is controlled by a computational unit 65, for example, a first order recursive filter. This may have the form of a single stage recursive filter such as that shown in FIG. 10. The input to this computational unit 65 is taken from a point 63 in the main part between the correlator input at 39 and the noise injection node 21. The computational unit 65 outputs a multiplier value that is dependent on the signal level measured at point 63 and is chosen such that the sum of the wanted signal coming from the limiter 15 and the added noise signal do not exceed the saturation level of any of the components that follow, specifically the summing node adder 21, the digital to analogue converter 19, the main amplifier 9 and the receiver transducer 11. The first order recursive filter 65 shown in FIG. 10 comprises a first unit 111 for measuring the absolute signal level. This is followed by a first multiplier 113 which produces at its output a signal that is one sixteenth of original level. This then is added by an adder 115 to a signal that has been delayed one cycle by a delay element 117 and scaled by fifteen sixteenths by a second multiplier 119. The output of this part of the first order recursive filter is then scaled by a preset factor of between one quarter and one sixteenth. The value of this preset scale multiplier may be fixed during manufacture or alternatively may be arranged to be preset by the user during initial set up. Accordingly the noise level can be preset to accomodate a user having either a poor or moderate hearing loss as appropriate.

By cancelling all of the effects of acoustic feedback and not simply inhibiting the onset and occurrence of oscillation, this new technique offers several user benefits:

(i) Inhibition of oscillatory (unstable) response;

(ii) Avoidance of perceivable "ringing effects" noticable at gain settings just below those that cause oscillation;

(iii) Removal of gain dependant perturbations in the frequency response which occur at frequencies far removed from the usual oscillation frequency.

As applied to hearing aids, the invention is intended to allow choice of a wider range of fittings. In particular it is intended to extend the range of availability of vented aids i.e. to allow such aids, particularly those having large diameter vents, to be prescribed where high gain is required in the higher range of audio frequencies.

We claim:

1. In a hearing aid comprising:

a microphone;

a preamplifier responsive to said microphone and producing a preamplified signal;

a main amplifier having an input supplied with an output electrical signal;

a receiver responsive to said main amplifier,

said receiver and said main amplifier having characteristics such that together said receiver and said main amplifier provide a substantially linear acoustic response to the output electrical signal when said output electrical signal has an amplitude within amplifier-receiver upper and lower linearity limits at the input of said main amplifier, and provide a substantially non-linear acoustic response to the output electrical signal when the output electrical signal has an amplitude outside said amplifier-receiver upper and lower linearity limits;

an electrical main pathway, connected between said preamplifier and said main amplifier, said electrical main pathway having a tapping point and a feedback insertion node;

an electrical feedback pathway connected across said electrical main pathway, said electrical feedback pathway extending from said tapping point to said feedback insertion node; and

filter means, inserted in said electrical feedback pathway, said filter means having characteristics which are calculated to model acoustic coupling arising between said receiver and said microphone;

whereby an electrical feedback signal shall be derived by extracting an electrical response signal from said electrical main pathway, passed via said filter means, and applied to said electrical main pathway to compensate for the acoustic coupling;

the improvement comprising:

amplitude limiting means, inserted in said electrical main pathway between said tapping point and said feedback insertion node, for limiting a residual electrical signal applied at its input to an amplitude lying within limiter upper and lower linearity limits in dependence on the amplifier-receiver upper and lower linearity limits so that the output electrical signal at the input of said main amplifier has an amplitude which is within said amplifier-receiver upper and lower linearity limits, and so as to suppress any non-linear acoustic response thereto which otherwise would result in an inadequate cancellation of the effects of acoustic coupling between said receiver and said microphone.

2. The hearing aid according to claim 1 wherein said filter means is an adaptive electrical filter, the hearing aid also comprising:

a noise source, connected to said electrical main pathway by a noise insertion node which is located between said tapping point and said feedback insertion node, to inject continuously an injected noise signal into said receiver; and

adaption control means, cooperative with said adaptive electrical filter, coupled to said noise source to receive said injected noise signal, and coupled to said electrical main pathway at a second tapping point located between said feedback insertion node and said noise insertion node for receiving the residual electrical signal formed of both the preamplified signal and electrical feedback signal as summed, and for correlating said injected noise signal and said residual electrical signal, thereby to adjust adaptively said adaptive electrical filter in dependance upon the electrical response of the hearing aid to the injected noise signal.

3. The hearing aid according to claim 2, wherein:

said amplitude limiting means is located in said electrical main pathway between said second tapping point and said noise insertion node; and wherein:

said limiter upper and lower linearity limits lie within said amplifier-receiver upper and lower linearity limits, and said limiter upper and lower linearity limits differ from said amplifier-receiver upper and lower linearity limits by an amount at least equal to the level of the noise signal.

4. The hearing aid, according to claim 2, wherein said preamplifier is followed by an analog to digital converter;

said main amplifier is preceded by a digital to analog converter; and

said noise source, said amplitude limiting means and said adaptive electrical filter are all digital.

5. The hearing aid, according to claim 4, wherein said adaption control means and said adaptive electrical filter comprise a signal correlator and a finite impulse response filter, respectively.

6. The hearing aid, according to claim 5, wherein said adaptive electrical filter has an adaption rate, and at least one of said noise source and said adaption control means has means for adjusting the adaption rate of the adaptive electrical filter.

7. The hearing aid, according to claim 6, wherein said noise source has means for adjusting the level of the injected noise signal to control said adaption rate.

8. The hearing aid, according to claim 7, wherein said means for adjusting the level of the injected noise signal is connected to said electrical main pathway and is responsive to the residual electrical signal so that the level of the injected noise signal is dependent upon the level of the residual electrical signal.

9. The hearing aid, according to claim 8, wherein said means for adjusting the level of the injected noise signal comprises:

a noise level control means connected between said noise source and said electrical main pathway; and

a first order recursive filter, connected to said electrical main pathway to receive the residual electrical signal, and connected to said noise level control means to control the level of the injected noise signal controlled thereby.

10. The hearing aid, according to claim 2, wherein the noise source comprises a pseudo-random-binary-sequence signal generator.

11. The hearing aid according to claim 4 wherein said noise source comprises:

a pseudo-random-binary-sequence signal generator; and

gate means connected both to an output of said pseudo-random-binary-sequence signal generator and to an output of said analog to digital converter for producing a randomized pseudo-random-binary-sequence noise signal.

12. The hearing aid, according to claim 1, including a molding configured to fit in the ear and having first and second passageways, said second passageway being open at both its ends to provide venting for the ear when said molding is fitted in the ear.

13. In a hearing aid comprising:

a microphone;

a preamplifier responsive to said microphone;

a main amplifier having an input supplied with an output electrical signal;

a receiver responsive to said main amplifier,

said receiver and said main amplifier having characteristics such that together said receiver and said main amplifier provide a substantially linear acoustic response to the output electrical signal when said output electrical signal has an amplitude within amplifier-receiver upper and lower linearity limits at the input of said main amplifier, and provide a substantially non-linear acoustic response to the output electrical signal when the output electrical signal has an amplitude outside said amplifier-receiver upper and lower linearity limits;

an electrical main pathway, connected between said preamplifier and said main amplifier, said electrical main pathway having a tapping point and a feedback insertion node;

an electrical feedback pathway connected across said electrical main pathway, said electrical feedback pathway extending from said tapping point to said feedback insertion node; and

filter means, inserted in said electrical feedback pathway, said filter means having characteristics which are calculated to model acoustic coupling arising between said receiver and said microphone;

whereby an electrical feedback signal shall be derived by extracting an electrical response signal from said electrical main pathway, passed via said filter means, and applied to said electrical main pathway to compensate for the acoustic coupling;

the improvement comprising:

amplitude limiting means, inserted in said electrical main pathway between said tapping point and said feedback insertion node, for limiting a residual electrical signal applied at its input to an amplitude lying within limiter upper and lower linearity limits in dependence on the amplifier-receiver upper and lower linearity limits so that the output electrical signal at the input of said main amplifier has an amplitude which is within said amplifier-receiver upper and lower linearity limits, and so as to suppress any non-linear acoustic response thereto which otherwise would result in an inadequate cancellation of the effects of acoustic coupling between said receiver and said microphone, said amplitude limiting means being a solitary limiter.

14. The hearing aid, according to claim 13, wherein said limiter has means for adjusting upper and lower limits of the amplitude of said residual electrical signal supplied to an input thereof.

15. The hearing aid, according to claim 14, wherein said limiter has means for externally adjusting the upper and lower limits of the amplitude of the residual electrical signal supplied to said input thereof.

16. In a hearing aid comprising:

a microphone;

a preamplifier responsive to said microphone;

a main amplifier having an input supplied with an output electrical signal;

a receiver responsive to said main amplifier,

said receiver and said main amplifier having characteristics such that together said receiver and said main amplifier provide a substantially linear acoustic response to the output electrical signal when said output electrical signal has an amplitude within amplifier-receiver upper and lower linearity limits at the input of said main amplifier, and provide a substantially non-linear acoustic response to the output electrical signal when the output electrical signal has an amplitude outside said amplifier-receiver upper and lower linearity limits;

an electrical main pathway, connected between said preamplifier and said main amplifier, said electrical main pathway having a tapping point and a feedback insertion node;

an electrical feedback pathway connected across said electrical main pathway, said electrical feedback pathway extending from said tapping point to said feedback insertion node; and

filter means, inserted in said electrical feedback pathway, said filter means having characteristics which are calculated to model acoustic coupling arising between said receiver and said microphone;

whereby an electrical feedback signal shall be derived by extracting an electrical response signal from said electrical main pathway, passed via said filter means, and applied to said electrical main pathway to compensate for the acoustic coupling;

the improvement comprising:

amplitude limiting means, inserted in said electrical main pathway, between said tapping point and said feedback insertion node, for limiting a residual electrical signal applied at its input so that the output electrical signal at the input of said main amplifier has an amplitude which is within said amplifier-receiver upper and lower linearity limits, so as to suppress any non-linear acoustic response thereto which otherwise would result in an inadequate cancellation of the effects of acoustic coupling between said receiver and said microphone; and

manual volume control means located between said amplitude limiting means and said feedback insertion node.

17. In a hearing aid comprising:

a microphone;

a preamplifier responsive to said microphone;

a main amplifier having an input supplied with an output electrical signal;

a receiver responsive to said main amplifier,

said receiver and said main amplifier having characteristics such that together said receiver and said main amplifier provide a substantially linear acoustic response to the output electrical signal when said output electrical signal has an amplitude within amplifier-receiver upper and lower linearity limits at the input of said main amplifier, and provide a substantially non-linear acoustic response to the output electrical signal when the output electrical signal has an amplitude outside said amplifier-receiver upper and lower linearity limits;

an electrical main pathway, connected between said preamplifier and said main amplifier, said electrical main pathway having a tapping point and a feedback insertion node;

an electrical feedback pathway connected across said electrical main pathway, said electrical feedback pathway extending from said tapping point to said feedback insertion node; and

filter means, inserted in said electrical feedback pathway, said filter means having characteristics which are calculated to model acoustic coupling arising between said receiver and said microphone;

whereby an electrical feedback signal is derived by extracting an electrical response signal from said electrical main pathway, passed via said filter

means, and applied to said electrical main pathway to compensate for the acoustic coupling;

the improvement comprising:

amplitude limiting means, inserted in said electrical main pathway between said tapping point and said feedback insertion node, for limiting a residual electrical signal applied at an input thereof to an amplitude lying within limiter upper and lower linearity limits in dependence on the amplifier-receiver upper and lower linearity limits so that the output electrical signal at the input of said main amplifier has an amplitude which is within said amplifier-receiver upper and lower linearity limits, and so as to suppress any non-linear acoustic response thereto which otherwise would result in an inadequate cancellation of the effects of acoustic coupling between said receiver and said microphone; and

the acousto-electric gain of said microphone, the gain of said preamplifier, the gain of said main amplifier, and the electro-acoustic gain of said receiver are all such that the hearing aid has an insertion gain which is matched to a predetermined prescriptive requirement.

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